Sound recording & reproduction

Glyn Alkin

This book is designed to bridge the gap between the professional recording engineer and the enthusiastic amateur who wishes to improve his techniques and gain a better understanding of recording medium and sound quality in general. Beginning with a look at the nature of sound itself, the reader is able to follow the progress of a recording from the acoustic environment through to the various forms of reproduction.

Sound recording is a highly complex process in which artistic and technical considerations are equally important. The author's long experience in the industry has convinced him that even the most artistic aspects of sound reproduction are capable of analysis given a thorough knowledge of the medium and of the programme requirements. However, he keeps theory to an essential minimum, concentrating on those areas that are of practical significance to people engaged in the art and science of sound recording.

For the reader who requires a more superficial approach at first reading, the easy-reference format and concise style make it easy to 'dip' without losing continuity. Subjects covered include acoustics, quadraphony, microphones, induction, sound control, stereophony, surround sound, theory of magnetism, magnetic recording, maintenance of reel-to-reel cassette and cartridge recorders, disc reproducers, noise suppression (Dolby) and digital recording.

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Sound Recording and Reproduction

Glyn Alkin

Focal Press · London & Boston

Focal Press
is an imprint of the Butterworth Group
which has principal offices in
London, Sydney, Toronto, Wellington, Durban and Boston

First published 1981

British Library Cataloguing in Publication Data

Alkin, Glyn
Sound recording and reproduction.
1. Sound-Recording and reproducing
I. Title
621.389'3 TK7881.4

ISBN 0 240 51070 4

American Library of Congress Catalog No: 80-41481

Phototypeset by Computer Photoset Limited, Birmingham
Printed In Great Britain by Mansell Bookbinding Ltd, Witham, Essex

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Introduction

This book is intended to bridge the gap between the professional recording engineer and the enthusiastic amateur who wishes to improve his techniques and gain a better understanding of the recording medium and sound quality in general.

The book follows the course of the production of a recording from the acoustic environment through to the various forms of reproduction, dealing with each subject in a practical manner and concentrating on those aspects that are of practical significance to people engaged in the art and science of sound recording.

Sound recording is a highly complex business, so the book includes some technical theory (where it is desirable for a better understanding of the medium) but this is tackled in as practical and non-mathematical a manner as possible. For those who only require a superficial approach (at least on first reading) the format, with a complete topic on every page, makes it possible to skip sections without losing continuity.

The Nature of Sound

Before starting to consider the process of sound recording, it is worthwhile pausing briefly to refresh our minds about the nature of the medium that we wish to capture.

What is sound?

Sound is an aural sensation stimulated by a vibration or mechanical wave motion in matter, operating within the frequency range that we can hear.

The simplest forms of sound vibrations, into which all sounds no matter how complex can be resolved, are pure tones such as can be produced by tuning forks. Tuning forks produce simple harmonic motions, i.e. motions which could be described by plotting the displacement of a swinging pendulum on a graph against time. The resulting graph is the familiar sine wave curve in which a complete cycle of vibration is indicated by an excursion both above and below the base line and return to the equivalent position.

How sound is propagated

Wave motion can be *longitudinal* (when the direction of motion is the same as that of propagation), *torsional* (a twisting motion) or *transverse* (as with the vibration of strings).

Sound waves require a material medium for propagation, eg air, water or wood. Unlike light, sound cannot travel through a vacuum.

When sound travels through the air, which is the way it usually reaches our ears (it is also possible to hear through direct mechanical contact by bone conduction), the wave motion is longitudinal. The waves consist of variations in air pressure, oscillating alternately above and below the prevailing barometric pressure. They travel outwards from the source at a speed of about 340 metres per second.

1130 ft. per second

Velocity of sound

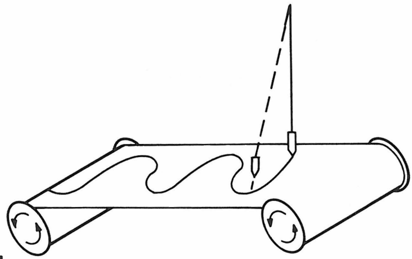
The actual velocity of sound waves depends upon the atmospheric pressure and the density of the air through which they pass. This is given by the equation:

$$v = \frac{1.41\ P}{D}$$

where v is velocity in cm/sec, D is the density of the medium in g/cm^3 and P is the barometric pressure in $dyne/cm^2$.

The velocity of sound increases with heat (which expands the air thereby reducing its density) and with humidity (since water vapour has a lower density than dry air).

Sound propagation through solids tends to be much more rapid and efficient than through air, for example, through steel, it can travel about fifteen times as fast.



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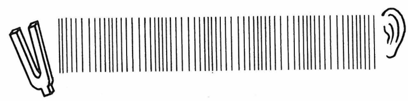

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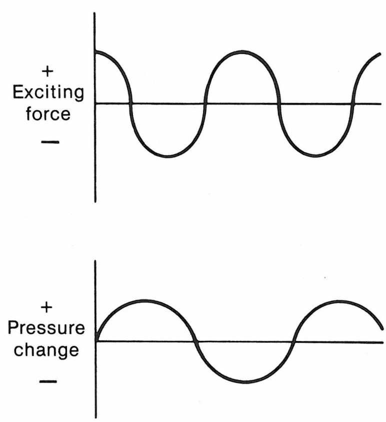

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1, An illustration of simple harmonic motion. If the paper is pulled with a smooth motion past a pendulam swinging in one plane, the trace will be a simple harmonic (sine wave) curve. If the movement of the tape is translated into position/time, the number of times that the curve repeats itself per second represents the frequency. 2, Sound is transmitted through the air by a series of alternating compressions and rarefactions, i.e. changes of air pressure above and below the prevailing atmospheric pressure; 3, Maximum pressure variations occur during maximum *change* in displacement of the exciting force, i.e. when it is crossing its normal state. Pressure variations are minimum when the force is at a maximum.

Sound waves are described in terms of amplitude, frequency and wavelength.

Sound Characteristics

Sound waves can be defined as those with a frequency range that lies within the audio spectrum.

Frequency

The rapidity with which a cycle of vibration repeats itself is called the frequency.

The audio frequency range is generally considered to extend between about 15 Hz and 20 kHz (one hertz (Hz) = one cycle per second). In practice the actual range of sounds that we can hear varies considerably between individuals, particularly at the high frequency end of the scale where aural sensitivity tends to diminish with age after about 25 years.

Wavelength

Having established that sound travels with a certain velocity and that each point of repetition of the waveform passes a given point with a certain frequency, it follows that these points must be a certain distance apart. This distance is known as the wavelength (λ). It is related to frequency (f) and velocity (v) by the simple equation:

$$\lambda = \frac{v}{f}$$

Many of the major problems in sound engineering stem from the wide variations in wavelength that comprise the audio spectrum, i.e. wavelengths ranging from under 2 cm to about 17 m.

Amplitude

The strength of a sound wave is known as its amplitude. This is a measure of the extent that the wave, ie the pressure change in air, deviates from the normal state at each instant.

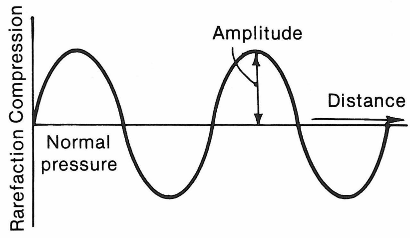
Loudness

When sound reaches the ear, the loudness we hear is not strictly proportional to the energy of the sound wave. The sensitivity of our hearing depends upon the frequency and the intensity of the sound.

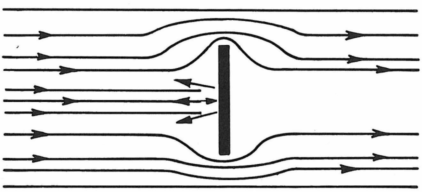
Just as our appreciation of musical pitch is determined by the ratio rather than the numerical value of the frequency interval, eg the frequencies at either end of an octave are always in the ratio of 2:1 regardless of actual numerical difference, so the difference in loudness depends upon the ratio, rather than the numerical difference in intensities. For this reason relative intensities are expressed in bels (B). This is a unit that compares the logarithm of their powers to the base 10. In practice the bel is too large a unit and the decibel (dB) is used. This represents about the smallest discernable change in audio power, equivalent to about 26%.

$$\text{decibels} = 10 \log_{10} \frac{P_2}{P_1}$$

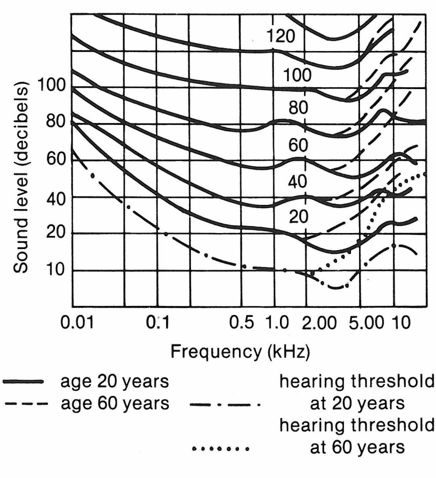
Sine wave curve representing the variations of pressure with distance for a pure tone.



Sound waves have a large range of physical dimensions comparable to many items of furniture and equipment etc. Sound is capable of bending around obstacles whose dimensions are smaller than the wavelength.



Contours of equal loudness plotted against frequency for pure tones at various intensities. The various curves represent different values in *phons.* The *phon* is a unit that takes into account the unequal sensitivity of the ear. It relates the intensities of sounds at various frequencies to an equally loud sound at 1 kHz. The solid line represents average sensitivities at age 20 years, the dotted line age 60. Note how our sensitivity to low frequency diminishes at low volume. This is the reason for tone-compensated loudness controls. The broken line at the bottom represents minimum levels of audibility.



Sound quality is subject to a number of different types of distortion which must be eliminated for high fidelity reproduction.

Sound Quality

The purpose of sound recording is to store information in such a manner that it can be retrieved later. Ideally the signal eventually reproduced should be exactly as recorded.

Frequency response

The recording/reproducing system should have a flat overall response over the desired frequency range—typically 30 Hz–15 kHz. At first sight (see table opposite), it would appear that the extreme high frequency range is unnecessary, especially as the main part of the musical scale and bulk of the audio power is in the middle/lower register. The high frequency range is important, however, as it contains the overtones, or upper partials, which give the various sound sources their characteristic timbre or quality. These overtones can extend to a very high order.

Harmonic structure

The difference between a particular note played on, for example, a piano or a violin depends upon the shape of the resulting waveform. Every waveform, no matter how complex, can be resolved into a fundamental (or lowest frequency), which determines the pitch of the note, and a series of overtones of higher frequency. The latter may be harmonics, ie multiples of the fundamental frequency, or unrelated inharmonics such as feature largely in the initial impact of percussion instruments. These starting transients are particularly significant in establishing the timbre, ie quality, of an instrument.

Amplitude response

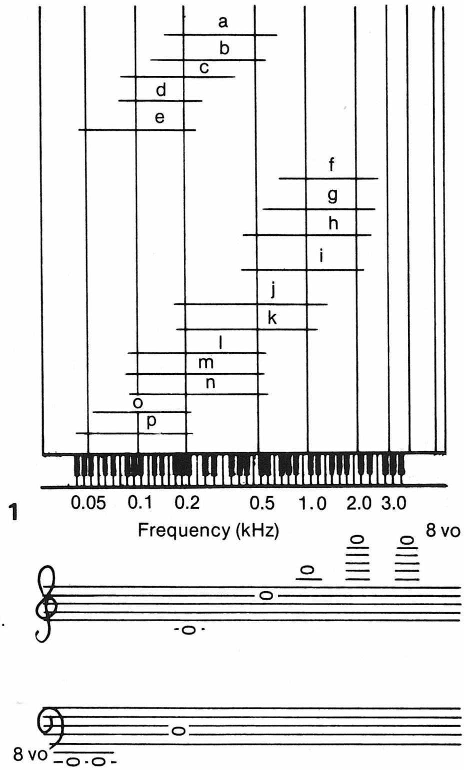
Variations in level of the input signal should be matched throughout by equal variations in the reproduced output otherwise *amplitude distortion* is present. An example of the deliberate use of amplitude distortion occurs in compression amplifiers where the variations in input level result in lesser variations in the output.

Linearity

Any non-linearity that occurs in the processing of the waveform, eg overloading, causing flattening of the curve can result in the production of spurious harmonics or *harmonic distortion.*

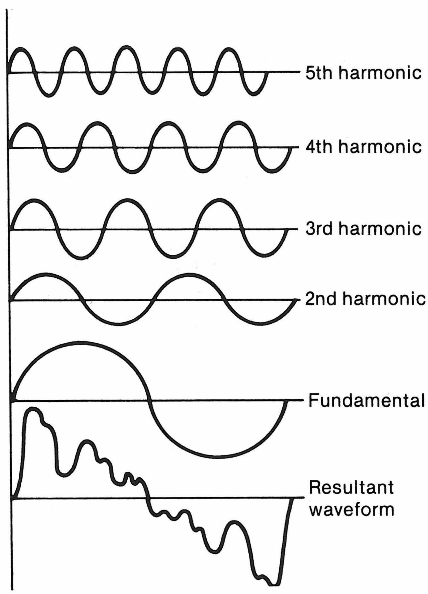
Intermodulation

If the various frequency components in a signal are caused to intermodulate with each other (multiply instead of add together) due to non-linearity in the system, *beat* frequencies (ie sum and difference tones) are produced which can form a long series of harmonics some of which will be discordant. This is known as *intermodulation distortion*. It creates an effect known as *acoustic roughness* which is found to be most objectionable when the beat rate (frequency difference) is of the order of 50 Hz for frequencies around 3 kHz.



1

The tonal range of voices and some musical instruments. a: Soprano. b: Contralto. c: Tenor. d: Baritone. e: Bass. f: Piccolo. g: Violin. h: Viola. i: Flute. j: Clarinet. k: Trumpet. l: Bass clarinet. m: Bassoon. n: Cello. o: Tuba. p: Double bass.
Only the fundamental frequencies are represented. The overtones extend throughout the upper audio frequency range.



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Analysis of a complex waveform into its fundamental and harmonic frequency components. The shape of the final waveform depends not only upon the number and strength of the individual overtones but also upon their relative phase, i.e. their time relationship to each other.

Listening via a microphone and loudspeaker can be very different from hearing the same sound direct.

The Hearing Process

Before considering recording processes in detail it is worthwhile to think about the nature of the material that is to be recorded.

For many people, their first attempt to record with a microphone is the first time that they come up against the problems of acoustics. They make recordings and are surprised to find how different they sound, even when the equipment used is technically immaculate. This can be largely due to the basic difference between listening in person and through a microphone. It is, perhaps, not generally realised that the combination of our ears and our brain provides us with an incredibly complicated mechanism that enables us to hear selectively.

The 'cocktail party' effect

If you are in the middle of a crowd of people, all talking to each other, it is usually quite possible to focus your hearing on a conversation that you find interesting, while effectively pushing all the other sounds into the background. You do not have to turn your head (although this helps) or move your ears like a cat. Your brain interprets the minute differences in time, volume and quality of the sound as it reaches each of your ears in turn. This gives you the ability to judge the direction of the sound sources and discriminate accordingly. On the other hand, microphones, although they may have directional characteristics, are not able to vary them from instant to instant. They pick up everything within their range, including all the reflected sounds that bounce back from the walls, floor and ceiling. Reproducing them in proportion through a loudspeaker robs the listener of his directional ability because all the sound comes from the same place.

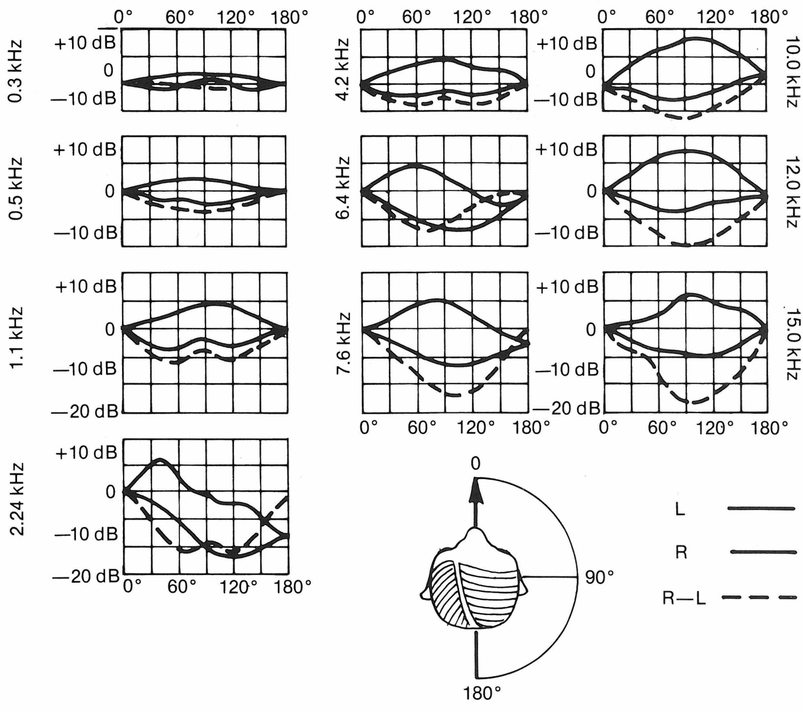
Binaural hearing

The ability of the listener to judge direction and discriminate between sources of sound can be restored to some extent with stereophony and almost completely with binaural reproduction (see pp 46 and 48). There still remains, however, such factors as the volume range normally being greater than is acceptable in the domestic situation and the inability of the eyes to direct the hearing and focus attention.

General factors

In general, the main differences between listening through a microphone and hearing direct are usually:

1. Greater awareness of the acoustic environment.
2. Less ability to distinguish between sources and less clarity.
3. Apparent increase in distance from the source.
4. Greater volume range (unless compression is applied).



We are able to judge the direction of a sound source because of:
1, The difference in time at which the sound reaches each of our ears; 2, The difference in sound pressure between them; 3, The difference in sound quality due to the acoustic effect of the shape of our ear lobes and head.
The diagram illustrates the way in which the sound reaching each of our ears differs with differing angles of incidence and at different frequencies.
It will be seen that there is little directional effect in the extreme low frequencies, marked difference in level above 6000 Hz and considerable difference in response at high frequencies.

When sound waves hit an object they are either reflected, refracted or absorbed.

Acoustics

On the previous page mention was made of the difference between listening in person and listening through microphones, one aspect of which is the *more obvious* effect of the acoustics. Let us consider what is meant by acoustics and their effect on the sound.

Reflection and absorption

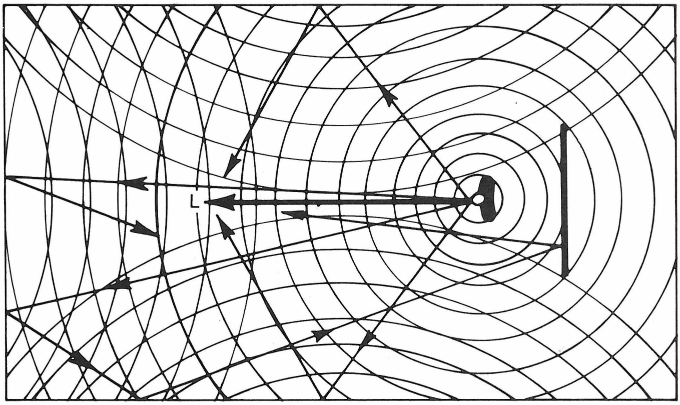
When a sound is made it radiates from the source in all directions, travelling outwards at approximately 340 m/sec until it meets an obstacle such as the walls, floor or ceiling. When it does so it is either wholly or partly absorbed or reflected, according to the nature of the substance it encounters. If the material concerned is porous, dense and of sufficient thickness, the sound waves will penetrate it and use up energy in finding their way through the 'labyrinth' of particles which therefore vibrate and expend energy in the form of heat.

The effect of wavelength

If the sound waves encounter a substance with a hard polished surface and if it is larger than the wavelength of the sound, most of the energy will be reflected in much the same manner as a mirror reflects light. Because of the very wide range of wavelengths involved and their comparable size with the objects they encounter, however, sound waves tend to suffer considerable diffraction and diffusion. Waves of longer wavelength than the dimensions of an obstacle will curve around it (especially if it has a streamlined shape) creating a small 'sound shadow' immediately behind the object but otherwise having little effect.

The walls, floor and ceiling of most rooms are reflective in their natural state. Plaster reflects about 97% of impinging sound and wood floors approximately 90%. Windows tend to be very reflective but this can be greatly reduced by curtains. The absorption characteristics of draperies depend on the density of the weave and the thickness of the material. Generally speaking this material will only absorb very high frequency sound. To reduce reflections at lower frequencies, draperies must be thick and, preferably, spaced away from the walls. In this way the lower frequencies (longer wavelengths) which would penetrate the material, use up their energy by attempting to make the sluggish mass vibrate.

In the same way, carpets and soft furnishings absorb sound but tend to have more effect at the top end of the frequency scale so that a room which is heavily furnished usually has a dull 'woolly' acoustic.



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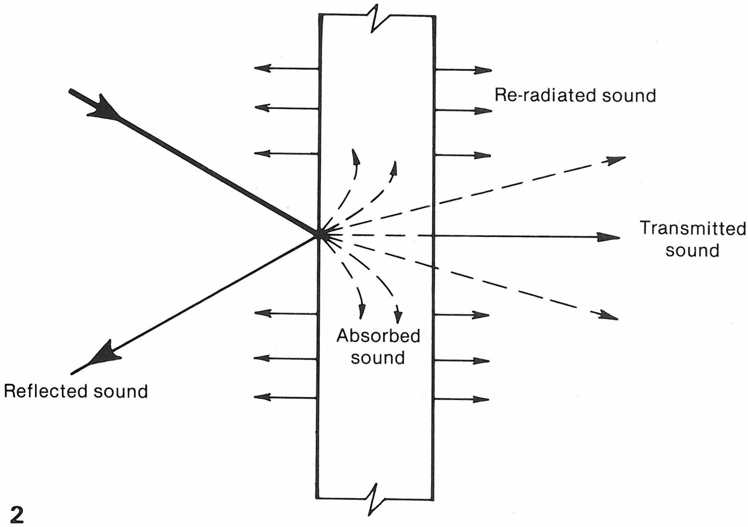


2

1, Sound radiates from the source in all directions. It rebounds from any reflecting surface larger than its wavelength and is refracted around surfaces that are smaller. It meets the listener (L) via a very large number of different paths; 2, When sound strikes a solid object such as a wall, some of the energy will be reflected, some absorbed. Unless the structure is very solid, some will be transmitted through it and some re-radiated by vibrations set up within it.

Sound energy in an enclosed space can build up and be sustained by multiple reflections, called reverberations.

Reverberation

When a sound is produced in an enclosed space, the waves will hit the walls, floors and ceiling and the proportion of their energy that is not absorbed in the surface will be reflected. The reflections will go on bouncing around the room, progressively losing energy on the way through friction with the air through which they travel and absorption at the various surfaces they encounter.

To a listener in the room, especially if listening via a microphone, these reflections will overlap and merge together so that the sound will be reinforced in volume and continue after the source has stopped. This is why people enjoy singing in a bathroom that has hard reflective surfaces (tiles) which provide effective reinforcement to the power of the voice. The overlapping reflections also tend to blur the diction and obscure minor detail in the performance.

Reverberation

The prolongation of sound by reflection is called reverberation (sometimes wrongly, called echo). It can be plotted for a given room and quantified as *reverberation time* in seconds or fractions of a second. Echo is a single reflection so delayed from the original sound that it is separately identifiable.

Reverberation time

The definition of reverberation time is 'the time that a sound, which has been cut off abruptly, takes to reduce to one millionth of its original power (ie to drop through 60 decibels)'.

Typical reverberation times are:

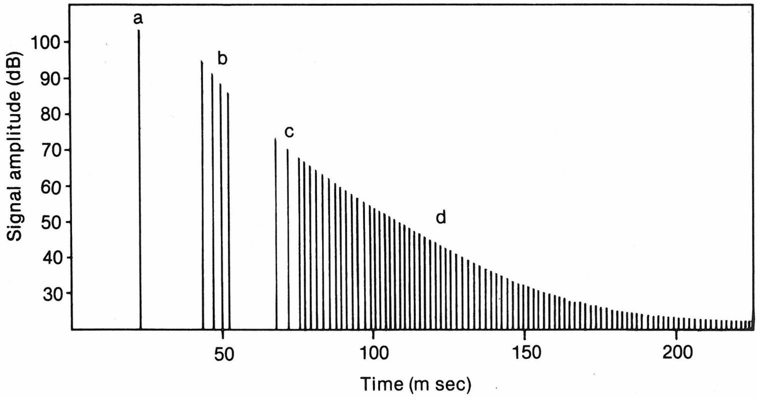
Average living room	0.6 sec	Large concert hall	1–2 sec
Small talks studio	0.4 sec	Large church	5–8 sec
Light orchestral studio	0.8 sec		

Decay curve

Just as important as the reverberation time is the manner in which the sound decays after it has been cut off. It is normal to assess studio acoustics by making an impulsive noise (eg a pistol shot) and plotting the shape of the decay curve with an oscilloscope measuring volume level against time. A good recording venue would have a smooth decay curve, reasonably equal throughout the frequency range, parabolic in shape and of appropriate length. Any humps in the curve would suggest standing wave reflections (eigentones) or ringing in the structure. These are to be avoided because they will produce a distorted, unclear recording.

In a small room standing wave effects can be mitigated by placing absorbent material between any parallel surfaces and not using microphones at the focus of reflective surfaces.

1



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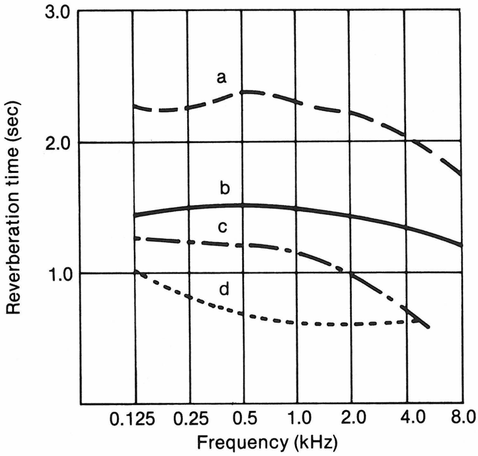

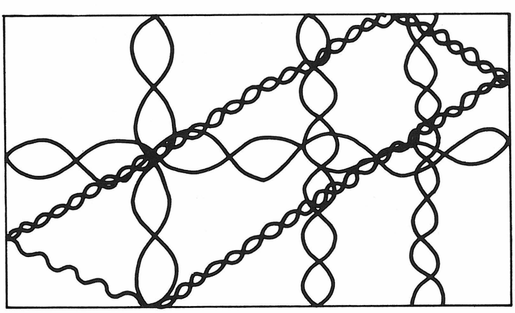

1, Illustrating the response of an auditorium to an impulsive sound: a: the initial sound; b: the first few reflections from comparatively close reflecting surfaces; c: reflections from nearest walls; d: multiple reflections from the body of the hall.
2, Typical reverberation times for various types of room and for various frequencies. a: very large hall; b: concert hall; c: large lecture room; d: living room.
3. Some standing wave patterns (eigentones) in a room.

The first requirement to produce a good recording is to choose, or arrange for, the correct acoustic environment.

Choosing the Right Acoustic

Appropriate acoustic

When it comes to assessing the suitability of a room or hall for recording it is important to relate the reverberation time (p. 20) to the size of the room and the type of programme material to be played. A long reverberation time, which would do much to enhance the blending of tones in well balanced classical music, could inhibit the acoustic/microphone separation necessary to achieve the right effect with a light orchestra that is not internally balanced. It could be quite useless for speech because the overlapping reflections could obscure the diction and make it unintelligible.

Acoustic compromise

Most recording studios have to accommodate a variety of different musical combinations and it is not unusual for the room to be much smaller than the ideal. It is, therefore, generally preferable to err on the side of too little reverberation in the studio. This will enable a number of microphones to be used with the minimum of mutual interference and will mitigate the 'boxy' effect that can be produced in a small room through the formation of standing wave patterns.

Performer comfort

The room must not be so 'dead' that it is oppressive to perform in. Anyone who has been in an anechoic chamber (a room with walls totally lacking in reflection used for testing microphones) will know just how oppressive this can be.

Where a quiet artiste is required to perform in a 'dead' acoustic or with a loud accompaniment (an obvious example is a vocalist with a 'pop' group backing), it may be almost impossible for him to hear himself so that his performance could suffer. In these circumstances it is usual to 'fold back' a portion of the microphone output to a loud-speaker close by, taking care not to allow the system to become unstable (howl-round), or provide them with headphones. However, the placing of reflective acoustic screens around individual performers or sections can help them to hear themselves as well as improving the sectional separation.

Noise

A very important factor in the choice of venue for recording is insulation from extraneous noise. The tolerable noise level depends upon the nature of the recording and the closeness of the microphone technique.

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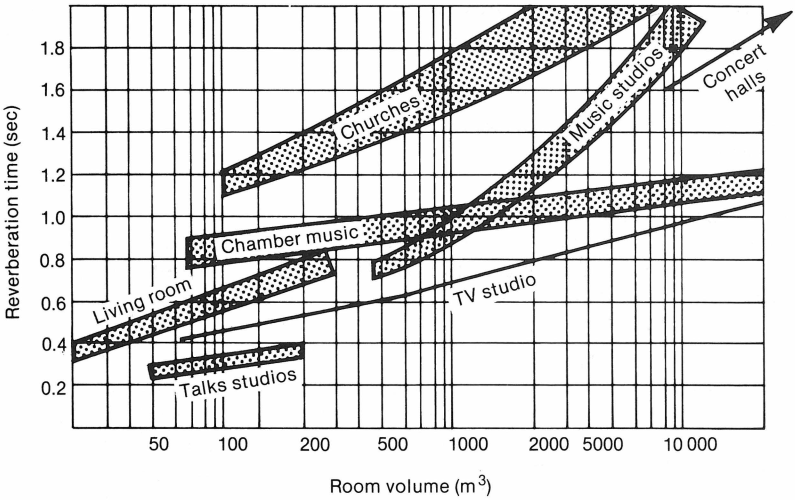

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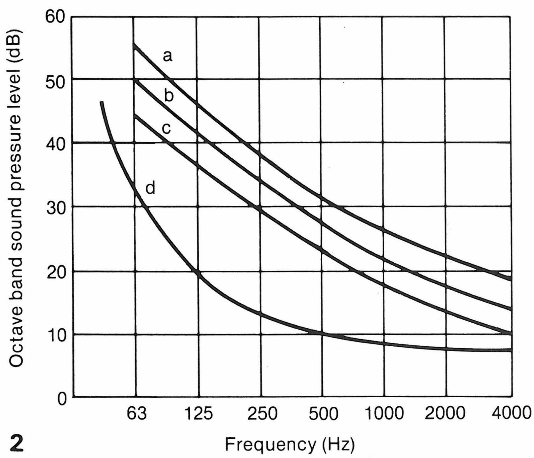


1, Recommended mid-frequency reverberation time for various sizes and functions of rooms. 2, Recording venues should be as quiet as possible. The table gives the accepted criteria for background noise levels in broadcasting studios. a: BBC audience studios; b: BBC television and sound studios (except drama); c: BBC sound drama studios; d: OIET sound studies.

Rooms to be used regularly for recording should have their acoustics measured and adjusted.

Acoustic Treatment

If a room is to be used only occasionally for recording, and particularly if a measure of acoustic variation is required, possibly the best arrangement is to provide a thick, felted carpet for the floors and thick velour drapes which can be drawn to cover any windows and at least two adjacent walls. These should be spaced about four inches off the walls.

Acoustic measurement

A room that is to be used frequently as a studio should, ideally, have its reverberation characteristics plotted for all frequencies and have proper acoustic treatment designed to match.

Absorption coefficient

The reverberation time can be measured by making a loud noise and measuring the time the sound takes to decay through 60 dB (see p. 20). Alternatively it can be assessed theoretically from a knowledge of the absorption coefficients of the various materials used in its construction. The absorption coefficient (α) of a material is the proportion of the incident energy arriving from all directions that is not reflected back. The following are some examples of mid-frequency absorption coefficients, α:

plastered wall	0.03
wood floors	0.1
thick pile carpet	0.5
curtains (medium) hung in folds	0.6

Sabine's formula

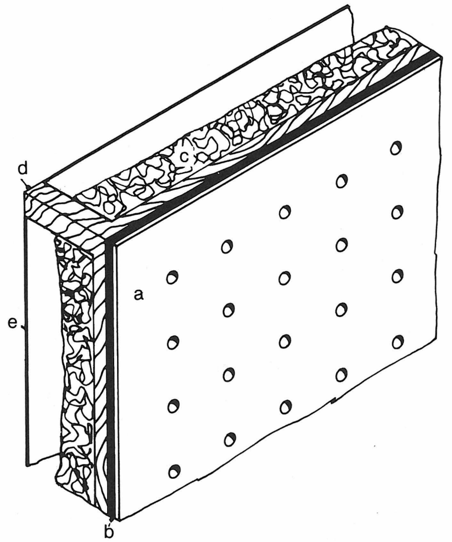
A fair approximation of the likely reverberation time of a room can be assessed by working out the mean absorption coefficient from the relative areas of the various absorbent materials and applying Sabine's formula:

$$\text{Reverberation time} = \frac{0.164V}{S\alpha} \text{ seconds}$$

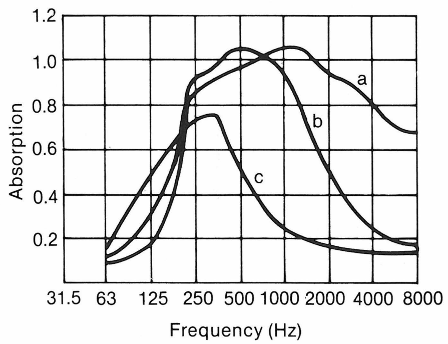
where V is the room volume (m^3)
S is the room surface area (m^2)
α is the mean absorption coefficient

The desirable reverberation time depends upon the size of the room and the purpose for which it is required.

Construction of a wide band absorber. a: perforated hardboard; b: bituminous roofing felt, bonded to the back of the hardboard; c: rockwool or glass fibre; d: timber frame; e: wall.
The rockwood or glass fibre absorbs the high frequency sounds which penetrate the holes and the roofing felt. The roofing felt damps the hardboard due to its inherent sluggishness and causes it to act as a damped membrane absorber. A wide range of absorptions can be arranged by making up boxes of differing sizes and thicknesses.



Typical absorption coefficients of wide band absorbers. The high frequency absorption can be determined by the ratio of perforation of the hardboard. The holes act as cavity resonators (Helmholtz resonators) with the air space behind. Curve a is for a 25% perforation, curve b for 5% and curve c for 0.5%.



Acoustic absorbers
Acoustics can be adjusted by panel absorption units. The most useful type is the wide band absorber that combines porous absorption for high frequencies with membrane absorption for the bass.

Modern recording technique requires the use of artificial reverberation.

Artificial Reverberation

It is most unlikely that the acoustics of the room or studio where a recording is to be made will be ideal for the purpose and, in any case, with modern microphone technique it is normal to apply differing amounts of reverberation to different sections of an ensemble. There is, therefore, much to be gained by starting with a studio with too little reverberation and adding it artificially where required.

Echo room

The simplest way to obtain artificial reverberation is to place a loudspeaker and microphone in a reverberent room, to feed a proportion of the required microphone output into the loudspeaker in the reverberation chamber (usually erroneously called the echo room) and then add a proportion of the reverberent sound picked up on the microphone back into the main output.

The reverberation chamber should have reflective walls that ideally are not parallel to each other otherwise standing waves will be formed. The path length between loudspeaker and microphone should be made as long as possible by the use of baffles.

Reverberation plate

A convenient method of obtaining artificial reverberation is the reverberation plate. It consists of a steel sheet approximately 2.5 m×1.5 m suspended vertically on springs. A moving coil transducer is fixed at a critical point near the centre and a piezo-electric contact microphone is mounted near one edge. When a signal is applied to the transducer a series of very complex vibrations radiate out to the sides and are reflected to and fro between the edges until they die out in the manner of reverberation.

The reverberation time can be adjusted by altering the spacing between the plate and another plate of similar size made of porous material held parallel to the first. The second plate effectively damps the vibrations in the reverberation plate.

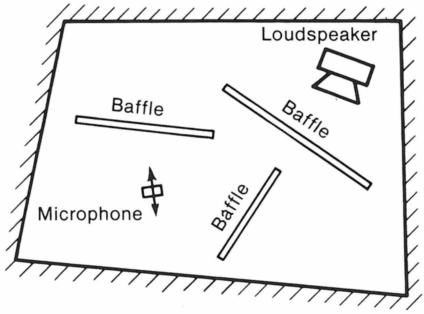
Reverberation springs

The most compact mechanical reverberation device consists of two sets of springs of different lengths joined together at the ends. Sound waves are induced into one end of the springs in a torsional manner and received at the other. The energy reflects back and forward between the two ends of the springs until it dies away, giving a reverberant effect. Another design uses a torsionally driven spring, providing a decay time of 2–4.5 sec.

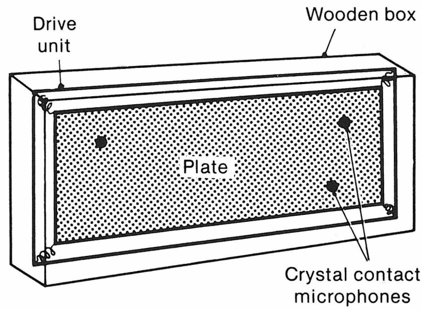
Microprocessor delay system

It is possible to build up a multi-delay device using microprocessors to simulate reverberation.

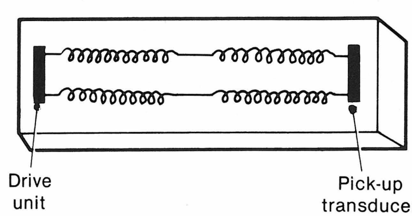
Plan of typical arrangement of a reverberation chamber (echo room). If possible the walls should not be parallel to each other. The baffles create a longer initial path between loudspeaker and microphones and help to break up standing wave reflections.



Reverberation plate. The specially selected metal sheet is suspended by springs at each corner in a wooden box. Vibrations are induced into it by a moving coil transducer and picked up by one, two (in the case of stereo) or four (for quadrophony) ceramic contact microphones.



Reverberation springs. Usually arranged in pairs, each with two springs of different lengths joined together.



The relationship between precedence and volume determines the apparent direction of a sound source.

Time Delay—The Haas Effect

There is a relationship between the order of precedence and the relative volume in which sounds are heard that determines the direction from which they appear to come. This is an important consideration in all multi-source sound systems whether they concern stereophony, surround sound or even a simple PA artist reinforcement system.

The Haas effect

Haas investigated the effect on the listener of two sources radiating the same sound from two different directions in the horizontal plane. These could be, for example, two loudspeakers fed with the same signal or a man and a PA loudspeaker reproducing his voice.

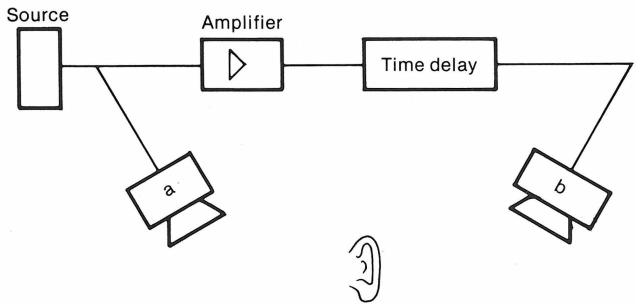
If the loudspeaker is nearer to the listener than the man (as often happens with PA systems), the sound from the loudspeaker will arrive first. If the sounds from the two sources have the same volume at the listeners' position, his impression will be that all the sound originates only from the loudspeaker and none from the man.

If a time delay of between about 5 and 50 msec is introduced into the feed to the loudspeaker, the impression will be reversed: none of the sound will appear to come from the loudspeaker although this contributes to the volume and reverberation of the sound. If, however, the volume of the delayed sound from the loudspeaker is increased relative to the direct sound from the man, a point will be reached when it overrides the time delay and the sound again appears to come only from the loudspeaker. The relationship between volume level and time delay in establishing the apparent direction of sound is illustrated by the graph opposite.

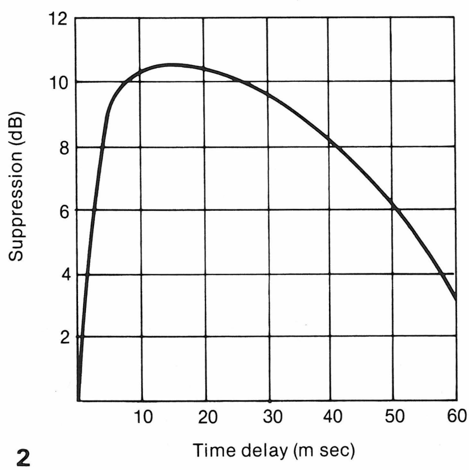
It will be seen that the maximum effect occurs with delays between about 10 and 30 msec. After about 60 msec transient sounds begin to be discernable as separate echoes. It should be remembered that the speed of sound in air introduces a natural delay of the order of 0.3 msec/m.

Artificial time-delay

The ability to insert time delay has enormous advantages, apart from the obvious one in the PA example quoted above. Time delay inserted in artificial reverberation systems improves realism, as it is normal for the first reflections to arrive about 50–100 msec after the direct sound. The ability to make sound appear to come from one direction when the bulk of it is coming from another has considerable artistic possibilities. Time delays can now be provided by digital means.



1



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1, If two similar loudspeakers (*a* & *b*) are fed from the same source one through an additional amplifier and variable time delay and listened to from an equal distance the following effects will be noticed: 1, If the loudspeakers produce the same volume, in phase, the sound will appear to come from a point equidistant between them. 2, If *b* is made louder than *a* all the sound will appear to come from *b*. 3, If the output of *b* is delayed with respect to *a* but the two volumes are the same, all the sound will appear to come from *a*. 4, The effect of the time delay can be overcome by increasing the amplification to make *b* louder than *a*.
2, The curve shows the relationship between volume and time delay in establishing the apparent position of a sound source. This shows, in the example of 1 above, that if 10 msec delay is inserted in the feed to loudspeaker *b* it would have to be over 10 dB louder than *a* to make the sound appear to come from a point between them.

Sound recording begins with a microphone, a device for converting acoustic energy into mechanical or electrical power.

Microphones

The aim of sound recording is to store mechanical energy derived from the minute movement of air particles caused by sound waves in a form that can be retrieved later.

History

The earliest form of sound recording consisted of a diaphragm which vibrated in sympathy with the sound waves connected to a stylus which ploughed a spiral groove in a wax cylinder. The depth of the groove varied with the diaphragm movement. The recording could then be played back by reversing the process, rotating the cylinder with the stylus in the groove so that the diaphragm followed the vibrations and radiated the sound. This mechanical system soon gave way to the electrical method of transmitting and recording sound which involves the use of transducers to convert acoustical energy into electrical impulses.

Microphones

A microphone is a transducer. It can convert acoustical energy into electrical power, either by direct contact with the vibrating source or, more usually, by intercepting sound waves radiating from a source.

Microphones that pick up sound from waves in the air have diaphragms to collect the acoustical energy. They fall into two basic categories according to whether one or both sides of the diaphragm are exposed to the sound waves.

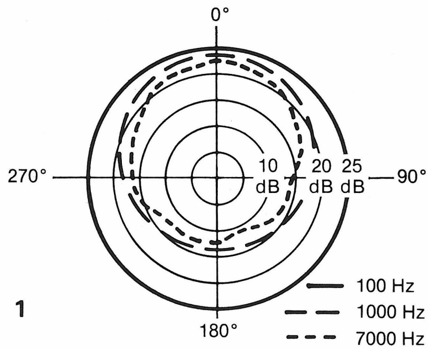
Pressure microphones

Pressure-operated microphones have only one side of their diaphragm exposed to the sound waves; the other side is sealed off except for a small aperture to allow the atmospheric pressure to equalise on both sides on a long-term basis (like the Eustachian tube in the ear). These microphones respond to minute variations in the surrounding air pressure. They therefore operate irrespective of the direction of the sound source unless their physical characteristics dictate otherwise (eg their dimensions are comparable with the wavelength of the sound) and are said to be omnidirectional.

Pressure gradient microphones

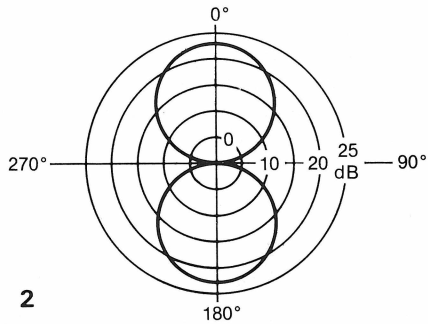
Microphones that have both sides of their diaphragms exposed to the sound waves operate by virtue of the *difference* in pressure (ie the pressure gradient) existing between the front and the back at any moment in time. This difference is a function of the different path length for the sound waves between the front and the back of the diaphragm. Obviously this path length difference will vary with the angle of incidence; this type of microphone, therefore, has a directional response.

The directional response of microphones can be represented by a polar diagram. This is a circular graph in which the centre represents the microphone. The distance from the centre of a line drawn around it represents the sensitivity at each angle with respect to its front axis.



1

1, A perfectly omnidirectional microphone would have a circular polar diagram. In the one depicted this is only true for low frequencies (100 Hz). At higher frequencies the 'shadow' of the microphone case reduces the output.



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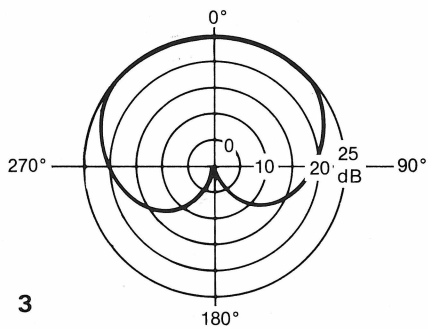
2, The figure-of-eight characteristic is typical of a ribbon microphone.



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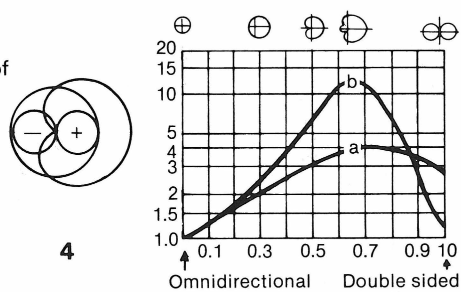
3, Cardioid (heart-shaped) polar diagram. The cardioid shape can be obtained by adding together equal omnidirectional and figure-of-eight outputs.



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4, Limacon curves show the effect of adding together different proportions of omnidirectional and figure-of-eight outputs.
Curve *a* gives the result in terms of directivity, ie the ratio of incident to ambient sound. Curve *b* shows the unidirectional factor, ie the ratio of axial response to response from all other directions.

There are many different types of microphones; the most popular are carbon, crystal, dynamic and capacitor.

Types of Microphones

There are four main types of microphones (discounting specialist types such as thermal and ionic transducers which are normally only used for scientific purposes). They are:

1. Carbon granule
2. Piezo-electric (crystal)
3. Dynamic (moving coil) and ribbon
4. Capacitor

The carbon microphone

In the carbon microphone sound pressure variations, acting on a diaphragm, are applied to a collection of carbon granules thereby varying their resistance to an electrical current due to an applied voltage. This type of microphone has a high output and a non-linear characteristic which tends to mute low level noise and limit when subjected to very loud sounds. It is thus very suitable for use in telephones but, due to its high noise level and variable output, is not often used for other purposes.

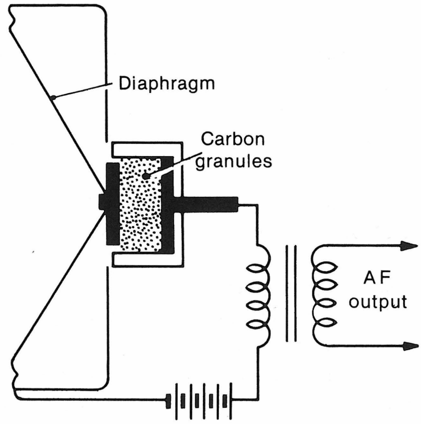
The piezo-electric (crystal) microphone

The crystal microphone works on the principle that certain crystalline materials such as quartz, tourmaline and rochells salts generate small electrical potentials when physically stressed. Various configurations are produced in which the movement of a diaphragm causes a wafer of the crystal to twist, producing a voltage across it. Some microphones have crystals joined together at the edges with an air space between them. This produces a higher sensitivity (of the order of 50 dB).

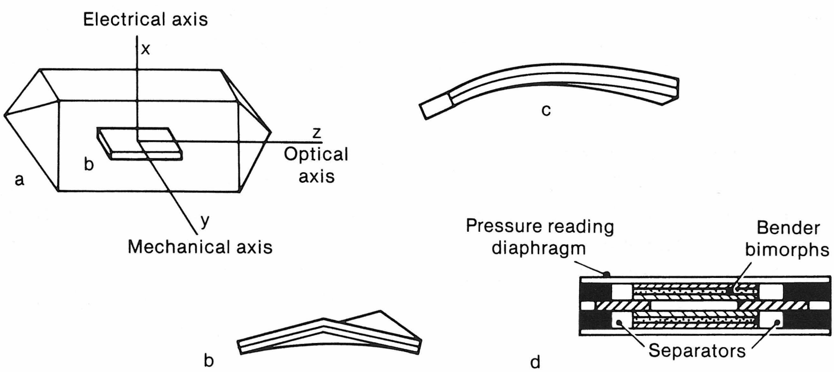
Crystal microphones have a very high impedance and need to be connected to an amplifier with an input impedance of at least 2 M Ω by a short, low impedance lead. A crystal microphone in fact functions like a voltage generator in series with a capacitor. If it is connected to an amplifier with too low an input impedance, the effect of its internal capacitors becomes significant and results in the loss of the high frequency response.

Transducers

Crystal elements also have many applications for directly converting mechanical movement into electrical voltages, eg gramophone pickups and contact microphones.



The carbon granule microphone. Sound pressure waves impinging on the diaphragm subject the carbon granules to varying compression. This alters their contact resistance and thereby the current through the transformer.



Crystal binorph element. Two bars are cut from X plates (*a*) on opposite diagonals and cemented together. Depending upon the way in which the crystals are cut, the resulting binorph can develop emfs between the crystal faces when the binorph is twisted (*b*) or bent (*c*). When the bar is bent, one crystal is under tension and the other compression so that emfs are induced in opposite directions. Two or more binorphs can be stacked on a box (*d*) with air gaps between to increase sensitivity. One face of the binorph can be used as the diaphragm of a microphone.

Moving coil microphones are robust, can have omnidirectional or unidirectional response and are of good quality.

Moving Coil Microphones

The moving-coil microphone consists of a diaphragm, which is usually a circular disc of aluminium alloy or plastic formed into a shallow dome with corrugations around the edge, close to where it is clamped, to allow it to move freely. To the back of the diaphragm is attached a coil of wire, usually wound from flat aluminium wire for lightness, which is suspended by the diaphragm in the annular gap between a powerful magnet and a central pole piece.

Induced voltage

When sound waves impinge on the front of the diaphragm, the back of which is partially or completely sealed off, the changing air pressure causes it to move in and out. This causes the coil, attached to the back of it, to move in the magnetic field, cutting the magnetic lines of force at right angles. As explained on p. 76, this induces an electromotive force, which constitutes the output, in the coil.

Microphone impedance

In order to reduce unwanted resonance of the diaphragm, the coil must be kept light and therefore consists of relatively few turns. Consequently the impedance of moving coil microphones tends to be low, characteristically about 30 Ω. Some incorporate a small step-up transformer into the stem of the microphone to increase the impedance to about 200 Ω. Some moving coil microphones have impedances that rise with frequency and need to be connected to amplifiers with an input impedance of about five times their own impedance to obtain a flat frequency response.

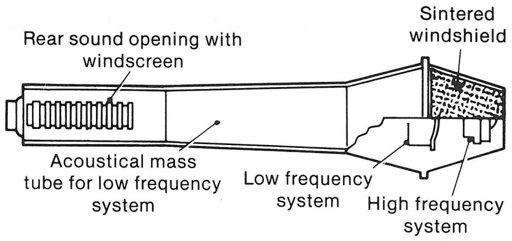
Omnidirectional dynamic microphones

Dynamic microphones that have the back of their diaphragms sealed from the air (except for a small pressure-equalising aperture) have an omnidirectional response for all frequencies with a wavelength longer than their dimensions. The larger types of microphone can create a 'sound shadow' which reduces the high frequency response off axis.

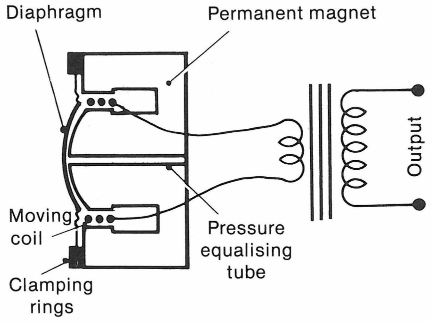
Directional dynamic microphones

Dynamic microphones can be made directional by allowing a proportion of the incident sound wave to reach the back of the diaphragm through some form of acoustic delay. This can consist of a tube to increase the path length or some form of acoustic labyrinth such as a fine porous material. The acoustic delay produces a difference in the relative pressures between front and back of the diaphragm which varies with the angle of incidence. Thus, by careful design, a cardioid (single-sided) response can be obtained.

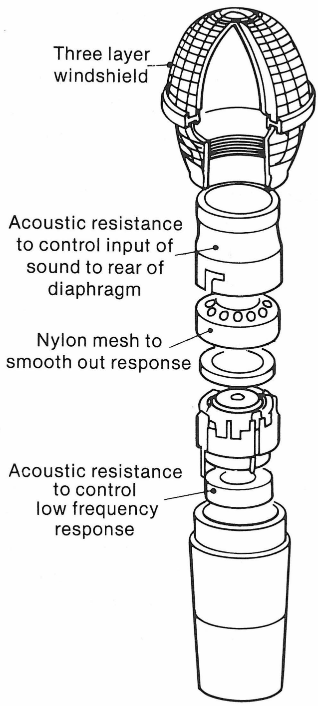
A double element directional microphone.
The problem of maintaining the proper phase relationship to achieve a cardioid response throughout the frequency spectrum is alleviated by dividing it into two with a separate transducer for each—one for the range 20–400 Hz and one for 400–18 000 Hz.



Simplified diagram of the construction of a moving coil microphone.
The diaphragm is dome shaped to promote stiffness.



Sectional view of a directional moving-coil microphone.
Sound waves approaching from the front axis of the microphone reach the front of the diaphragm direct and the back via an acoustic delay so that a phase difference is created. Waves from the back encounter the acoustic resistance first, arrive at both sides of the diaphragm in the same phase and cancel.



Capacitor microphones can be small and compact and have excellent response characteristics.

Capacitor Microphones

The electrostatic (capacitor) microphone is really a small variable capacitor in which one plate is formed by the backplate and the other by the diaphragm, which is evenly spaced very close to it. Characteristically, the distance between the plates is about 1–1½ thou' giving a capacitance of about 10–20 pF.

Variations in sound pressure cause the diaphragm to move with respect to the backplate so that the capacitance varies in relation to the sound wave.

Polarising the capacitor

In order to obtain an output the variable capacitance must be turned into an emf. This is done by applying a polarising potential of about 60 V through a high resistance (100 MΩ or more). Variations in capacitance caused by movement of the diaphragm are thus converted into voltages across the resistor which is then applied to a head amplifier (usually consisting of a field effect transistor) contained in the stem of the microphone.

Electret diaphragms

Some capacitor microphones employ electret diaphragms. This is a material (usually a high polymer plastic film) which has been given a permanent electrostatic charge. The need for a polarising voltage is thus eliminated and the whole assembly can be made much more compact. This is a very useful feature for 'lavalier' microphones which are worn on a lanyard around the neck or on a tie clip.

RF capacitor microphones

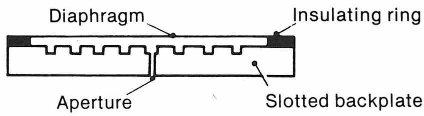
Another type of capacitor microphone uses the variation in capacitance to beat with an RF oscillator, the rectified signal comprising the audio frequency output. This type of microphone can have an excellent frequency response and good sensitivity but can produce variable results when subjected to large variations in temperature and humidity.

Directional response

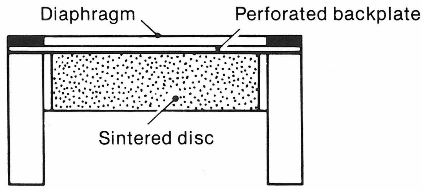
Capacitor microphones can be operated by pressure and are therefore omnidirectional if the back of the diaphragm is sealed (except for a small pressure equalising path). They can be made into very small capsules which do not produce sound shadows and thus retain their all-around response at all frequencies.

Alternatively the backplates can be made of a porous material which forms an acoustic delay path and produces a unidirectional response.

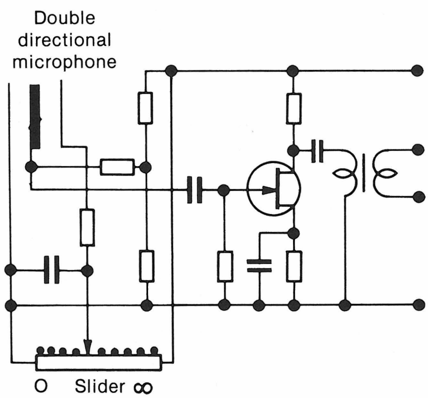
Sectional view of a pressure-operated capacitor microphone. The solid backplate is slotted to reduce the back pressure behind the diaphragm. The diaphragm is made of thin aluminium or metalised plastic stretched over an insulating ring. There is a small aperture to allow the pressure to equalise on a long-term basis.



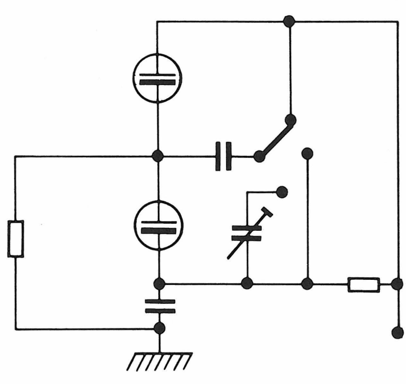
Sectional representation of the essentials of a pressure-gradient (directional) capacitor microphone elements. The sintered disc forms an acoustic impedance.



A method of achieving variable directivity using a double-directional microphone, ie two diaphragms separated by a shared porous back plate. Moving the slider determines the relative polarity between the plates and produces a range of directivity patterns continuously variable between omnidirectional and figure-of-eight patterns.



A method of connecting a double-element microphone (one omnidirectional and one cardioid) to give three different switched responses.



Personal Lavalier or Clip-on Microphones

There are several basic methods by which sound can be picked up from a person who is moving about.

Microphone boom

From the sound quality point of view, mobile sound sources are best dealt with by suspending the microphone on a telescopic boom or on a smaller 'fishing rod'. These can be moved about to follow the artist while maintaining the correct relationship between them.

Hand-held microphones

Hand-held microphones must be robust. They are usually provided with close-talking windshields and built-in filters to reduce the bass boost and possibly high frequency sibilance caused by close working.

Lavalier microphones

Miniature microphones, which need not be held, can be hung around the artist's neck by a lanyard (lavalier) or clipped to the clothing. These microphones can be either crystal, miniature moving-coil or, in the case of the better-quality ones, electrostatic.

The electrostatic microphones generally use electret (permanently polarised) diaphragms to eliminate the need for a polarising voltage supply. They do, however, need a head amplifier and this is usually arranged in the form of a small cylinder which can be housed in a pocket and connected to the microphone by a short length of thin flexible cable. The connection to the amplifier is usually a much heavier cable which can stand up to being dragged along the floor.

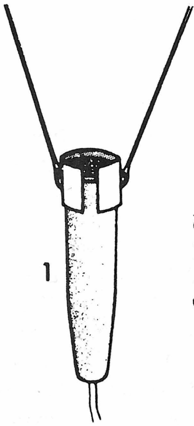
This type of personal microphone can produce quite good results, provided that it is carefully positioned on the person and is not shielded by the clothing. It is, however, an unnatural position from which to pick up the voice; there may be bass boom (due to chest resonance), excessive nasal tone and lack of sibilants as the nose is in line with the microphone which may be shielded from the mouth by the chin. This can give the sound a distant 'off mike' quality. For these reasons, some personal microphones are designed with 'presence' peaks and a diminishing bass response, others have external filters to modify any flat characteristic and, in one type, the

adjusting response

response can be adjusted by altering the position of the lanyard clip which forms a cavity resonance.

Wireless microphones

Both hand-held and lavalier microphones suffer from the disadvantage that they employ a cable which can restrict the artist's movement. Radio transmitters are available of about the size of a cigarette package into which personal microphones can be connected (also eliminating the need for head amplifiers in the electrostatic type). Frequency modulation is used, the choice of frequency depending upon the presence of interfering signals in the vicinity.



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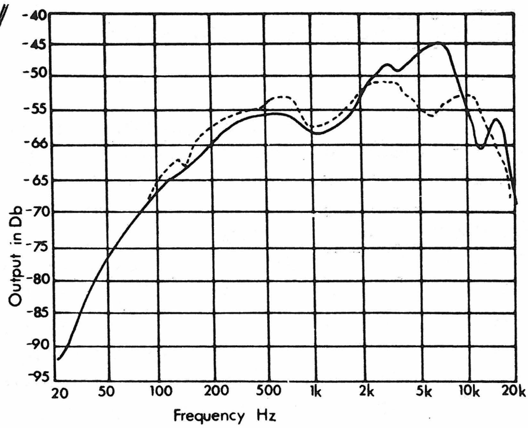

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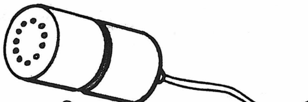

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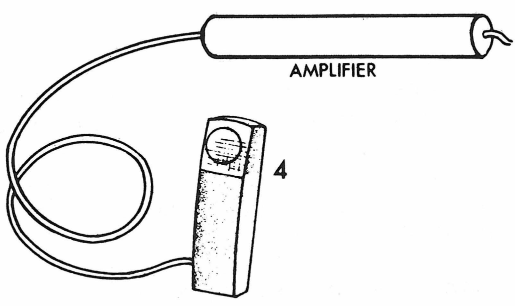


4

MICROPHONES

Personal or lanyard

A personal lanyard microphone incorporating a movable clip. 2, Illustrating the effect of moving the clip. When the clip extends over the top of the microphone a cavity resonance effect causes an increase in the response at around 6 KHz giving increased 'presence'. 3, A miniature electrostatic personal microphone which uses an electret diaphragm. 4, An electrostatic microphone in which the diaphragm faces away from the artist wearing it. The idea is to give preference to the other person in an interview situation.

Ribbon microphones can have excellent response characteristics but are prone to handling noise and wind blasting.

Ribbon Microphones

In the ribbon microphone the diaphragm consists of a narrow ribbon of corrugated metal foil stretched between the pole pieces of a powerful magnet.

Pressure gradient operation
The ribbon is exposed to the sound waves on both sides. It is activated by the difference in sound pressure between the front and the back of the ribbon at any instant in time. Sound waves approaching from the front of the microphone take longer to reach the back and vice versa so that a difference in pressure is created between the front and the back of the ribbon, which causes it to move away from the direction of higher pressure.

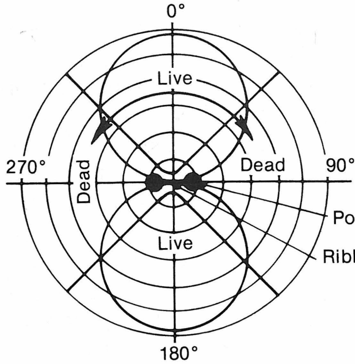
Electromagnetic induction
As in the moving coil microphone (p. 34) the movement of the conductor (in this case the ribbon) in a magnetic field induces an electric current into it. This is applied to a step-up transformer (to increase the impedance) to provide the output.

Directional characteristics
As the operation of the ribbon microphone depends upon the difference in path length between front and back it follows that it will have maximum sensitivity at the front and back and zero output at right angles where the path length to each side of the ribbon is identical.

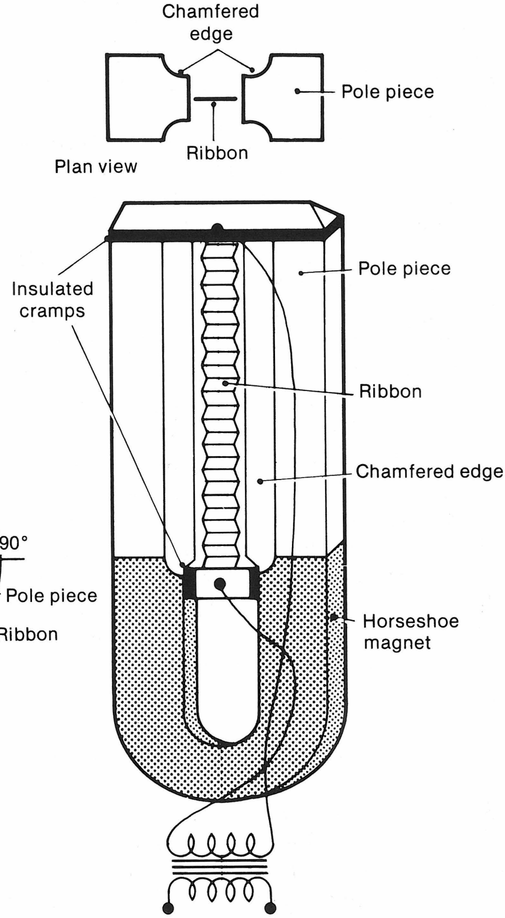
Polar response
In a polar response curve, in which the sensitivity of the microphone in any direction is represented by the distance of a line from a point representing the microphone, the microphone will have a figure-of-eight configuration. This means that the microphone can be considered to be live over an angle of approximately 90° at the front and back and dead at the sides.

Unidirectional ribbon microphones
Ribbon microphones can be given unidirectional characteristics by blocking off the back of the ribbon with an acoustic delay. This matches the two path lengths so that sound approaching from the rear reaches the front and the back (through the delay) simultaneously and cancels out.

General characteristics
Ribbon microphones can have excellent frequency response characteristics but they have a small, low impedance output, tend to be delicate and suffer from handling noise and wind blasting.



Polar response curve



The ribbon of a ribbon microphone is made of very thin corrugated aluminium alloy. It is held by two insulated cramps between the pole pieces which are extensions of the powerful horseshoe magnet.

Sound has equal access to both sides of the ribbon and high frequencies are also able to pass between the gaps at the sides of the ribbon. To offset the resulting loss the pole pieces are usually chamfered (e). This creates an acoustic resonance which boosts the HF response.

Ribbon microphones have figure-of-eight polar characteristics in both the horizontal and vertical planes but they tend to be lacking in HF response off axis in the vertical plane when the wavelength of the sound becomes comparable with the length of the ribbon, causing a wave motion along its length.

Microphones for long distance pick-up can be based on the 'collection' or 'rejection of sound' principle.

Super-directional Microphones

There are numerous applications where microphones have to be used at a considerable distance from the sound source. These can include: television and radio production (where microphones are often required to be kept out of the picture), stage sound reinforcement and nature recording.

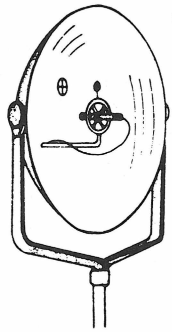
The requirement is for a microphone with a very narrow acceptance angle to give maximum discrimination between the wanted source and the ambient noise and reverberation.

1. The larger the reflector, the more of the pressure wave it will collect and scoop into the microphone and thus the greater the forward sensitivity.
2. Surfaces can only reflect sounds of wavelength shorter than their dimensions (see p. 18) so that for a parabolic reflector to operate throughout the full audio frequency range it would apparently need to be of the order of 30 ft in diameter! In practice, where the microphone is used close to a reflecting surface, a pressure-doubling effect takes place which tends to preserve the action down to a lower frequency than the dimensions would suggest.

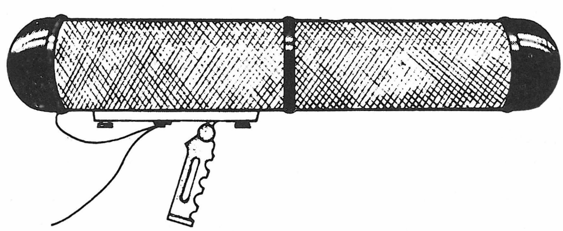
Practical parabolic reflectors are usually between 18 in and 3 ft in diameter. The ideal arrangement is to use a microphone with a uni-directional response just wide enough to embrace the whole of the dish. The microphone faces towards the dish and away from the source of sound so that it only receives reflected waves which are in phase. Sound reaching the microphone direct can be out of phase and cause cancellation with the reflected sound. A 3 ft parabolic microphone can provide considerable sensitivity down to about 250 Hz, below which the amplifier gain should be 'rolled-off'.

Gun microphones

Even an 18 in diameter parabola is a cumbersome piece of equipment. The search for other forms of narrow-angle microphone has led to the development of the 'line' or 'interference tube' microphone, usually called the 'gun mike'. Unlike the parabolic reflector, this works by rejecting unwanted sound rather than collecting sound from the wanted direction. The microphone element is coupled to a tube, typically about 6 in, 18 in or 3 ft long (one example is 6 ft long), with a series of holes or a slit running down its length. The theory is that sound arriving at the microphone along its axis passes straight down the tube with little effect, whereas sound arriving from other directions enters the holes or slot at differing points along the length, travels differing distances to the microphone and thereby tends to cancel out. A finely graded acoustic impedance is usually incorporated in the tube to even-out the pressure waves from different parts of the tube and to discourage sound entering one part of the tube merely emerging from another.



unidirectional microphone faces dish



Top: Microphone in parabolic reflector. The unidirectional microphone should be facing into the dish and away from the source of sound. Note sight hole and cursor for aiming the dish. *Middle:* Rifle (interference tube) microphone. Note sound entry slits down one side. *Bottom:* Rifle microphone with windshield and pistol grip.

There are no hard and fast rules about the choice and positioning of microphones but here are some general points.

Using Microphones

How many microphones?

The golden rule about the number of microphones to use is *the fewer the better* because, unless the source for each microphone is isolated to it, the overlapping sound fields will reduce the clarity of the sound. Nevertheless, the fact remains that there are many circumstances where the desired quality of sound can only be achieved by multi-microphone technique. At one end of the scale it is possible, given the right conditions, to record a symphony orchestra of 80 musicians on one stereo pair, while at the other there are 'pop' groups that use 10 microphones on a drum kit. The factors governing the number of microphones are: the internal balance of the source; the closeness of the technique required to produce the desired quality of sound; and, whether different elements in the balance require different degrees of processing, ie frequency response shaping, reverberation, etc.

Internal balance

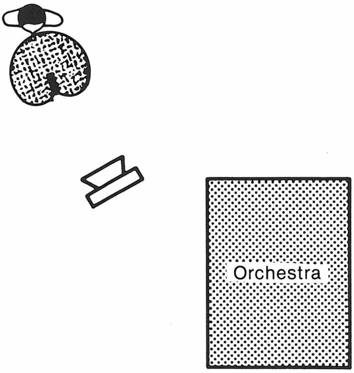
Very few programme sources are sufficiently balanced within themselves for recording purposes, bearing in mind the limitations of reproduction in the domestic situation (see p. 16). Even in the case of the symphony orchestra, mentioned above, the conductor would expect to receive guidance on points of musical balance from the sound balancer because, standing so close to the orchestra, it is not possible to assess the balance between the musicians with sufficient accuracy for recording purposes. The majority of musical combinations are not at all well balanced and require each section or musician to have a separate microphone which can be separately controlled to obtain a balanced mixture at each moment in time. This multi-microphone technique throws considerable responsibility on the balancer but in the professional recording business they tend to be musicians, specially trained for the purpose.

Sound character

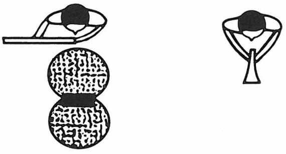
The character of the sound depends upon the type of microphone, its proximity to the source and the technical processing applied to its output. The main factors governing closeness are the ratio of incident to ambient sound, ie separation from reverberation and other sources, and sound quality. Most instruments produce differing frequency characteristics from different parts and in different directions. At varying distances these frequencies suffer differing attenuation and phase relationships. To obtain the sort of crisp attack that is demanded of modern music (reverberation can be added artificially later) and achieve good separation between the instruments a close technique is necessary. This in turn demands multi-microphone technique because it is not possible to get close enough to a number of sources simultaneously.

A hypercardioid microphone to give maximum discrimination against PA loudspeakers.

A cardioid microphone for a vocalist angled so as to discriminate against the orchestral accompaniment. The angle of discrimination of a cardioid microphone is much more acute at the back than at the front. The small 'fold-back' loudspeaker may be necessary to enable the vocalist to hear himself.



The figure-of-eight characteristics of a ribbon microphone used to isolate the comparatively weak sound of a flute against the powerful tones of a trumpet.



In selecting the type and position of a microphone for a particular purpose it should be remembered that it is often not so important to point the live side of the microphone directly towards the wanted source as the dead side towards the source to be discriminated against.

When using more than one microphone it is important that they are properly phased. This can be checked by placing them near to each other and speaking between them. If they are out of phase the sound will become distorted and weaker when they are both faded up.

Receiving separate information in each ear enables us to locate the direction of a source of sound.

Binaural Hearing

So far we have only considered monophonic sound, as though listening with only one ear. Most of us, however, are able to listen with two ears and this gives us the ability to judge the direction of a source of sound and, to a certain extent, to discriminate between sounds coming from different directions.

When a sound comes from a direction other than straight ahead it reaches the nearer ear slightly before the other one. The brain is able to detect this tiny difference in time and also slight difference in intensity and quality (due to the waves being defracted around the head). It can interpret these to gain information about the direction of the sound source. We can also tell whether the sound is coming from in front or behind by virtue of the slight difference in quality that the shape of our ear lobes imparts to sound from each direction. Discrimination in the vertical plane can also be achieved to a limited extent due to the ear lobes and the fact that we seldom hold our heads quite still and exactly vertical.

Aural discrimination

The ability to locate the direction of the various sound sources can help us to concentrate our attention on wanted sounds and subjugate unwanted sounds such as noise or reverberation, etc. If this ability can be applied to sound recording, it can greatly enhance the realism and clarity of the reproduction.

Binaural reproduction

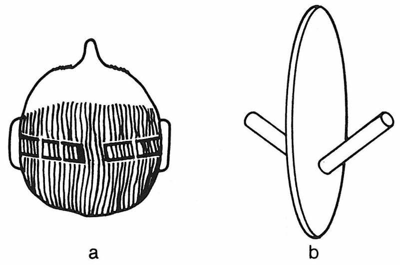
The nearest approach that can be made to the binaural experience by electrical means involves placing two microphones one in each side of a dummy head which is designed to reproduce as closely as possible the acoustic proportions of the human head.

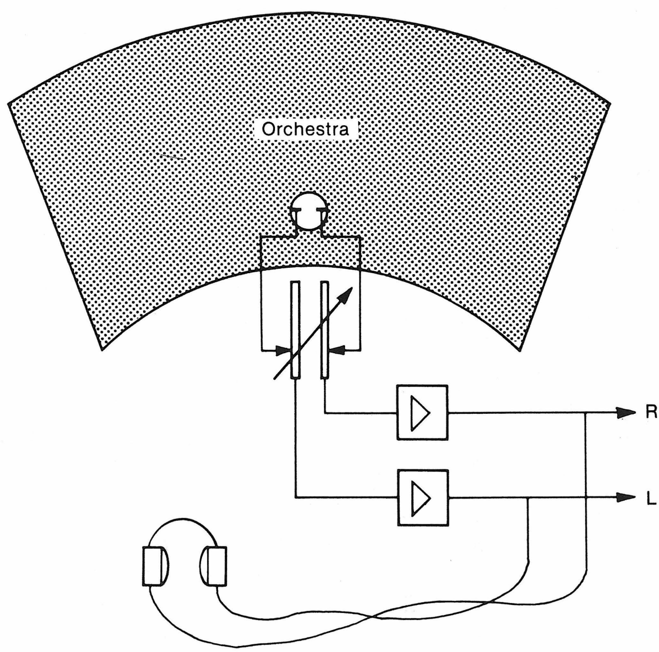
The outputs of these two channels must then be kept separate (though equal in phase relationship) throughout the entire process of recording and/or transmission and reception to end up each in the appropriate earpiece of a pair of headphones.

The resultant effect can be very realistic, provided that the above 'purist' microphone technique can be employed. It is much less suited to the method of multi-microphone balancing adopted for most stereo recordings. A sound source 'panned' electrically into the middle (see Stereophony) can appear to emanate from the middle of ones' head in a rather disturbing manner.

A schematic illustration of binaural sound reproduction.

A pair of microphones is mounted inside a dummy head, usually made of polystyrene, to copy the appearance of a human head as far as possible. Alternatively, the pair of microphones can be mounted at an angle (usually 90–180°) with a perspex baffle of approximately 30 cm diameter between them.





The microphone arrangement is placed in a position where the internal balance is optimum, eg above the conductor's head in an orchestra.
The two outputs must be kept separate, but in phase, throughout the recording/broadcast process so that they can be connected to the respective earpieces of a pair of headphones.

Stereophony

On the previous page we discussed the realism that can be added to reproduction by listening binaurally, ie with two ears. There are two disadvantages in the system, however:

1. It requires the listener to wear headphones, which is not always convenient.
2. It requires a microphone technique that is only suited to sound sources that are well balanced within themselves, eg symphony orchestras.

Stereophony, on the other hand, is a two-channel sound system that is intended to be listened to via loudspeakers.

The basis is rather similar to the binaural system in that it sets out to provide the ears with separated sources and thereby supply information about the direction of the sound. Because loudspeakers are used, however, the separation is not complete. It is therefore common practice to make the positional information rather larger than life so that it is still effective even when diluted by the acoustics of the listening room.

Stereophonic microphone technique

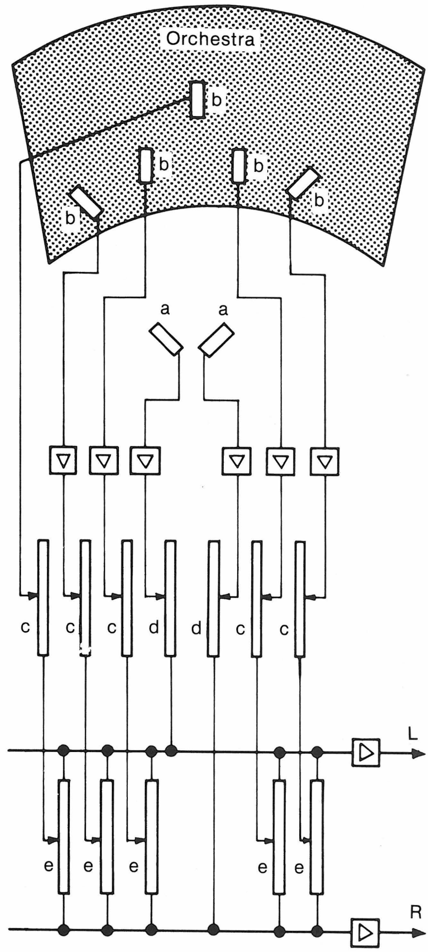
A typical approach is to employ a pair of microphones close together, one aimed 45° to the left and one at 45° to the right of the sound source (or possibly 90°, ie 180° to each other) to provide an overall 'audio picture' of the subject, eg an orchestra. The outputs of a number of 'spot' microphones positioned closer to the sound sources are then added. These are carefully mixed into the overall output to provide each source with the necessary degree of accentuation for balance purposes. The outputs can also be 'pan-potted' into an apparent position which, if an overall stereo pair of microphones is used, should agree with their geographical location.

Pan pot

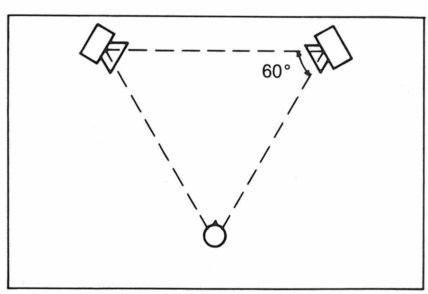
The term 'pan-pot' is derived from two words, panorama and potentiometer. The verb 'to pan' comes from the motion picture makers' expression meaning to change direction in the horizontal plane. The pan potentiometer is a resistance virtually connected between the left and right channels with a slider to which the microphone output is connected. Moving the slider along the resistance adjusts the proportion of the microphone output supplied to each channel and thereby appears to 'pan' it across the space between the two loudspeakers.

Stereo listening conditions

To hear stereophony properly, the two loudspeakers should be perfectly matched, equally balanced for volume and in phase. They should be at least 2 m apart, the listener forming an isosceles triangle.



A schematic diagram of an orchestra with a microphone balance consisting of a coincident pair of stereo microphones (*a*) and a spot microphone (*b*). Each microphone is adjusted for level by the channel faders (*c*) and the 'ganged' pair of faders (*d*). The outputs of the spot microphones are then adjusted by means of the pan pots (*e*) to retain a geographical relationship that agrees with the positional information in the output of the stereo pair.



For effective stereophony the listener should be equidistant between the loudspeakers and at an angle of about 60° to each.

Quadraphony provides sound with positional information for all four quadrants around the listener.

Quadraphony

The limitations of stereophonic sound

Stereophonic sound, discussed on the previous page, can provide the listener with positional information about the sound source. It can, in effect, present an aural picture of, for example, the orchestra spread out across the room in the space between the two loudspeakers. This can greatly enhance the realism of the reproduction and give the listener a measure of choice in balancing the various sources (ie instruments) and balancing them and the studio ambience. In this last respect, it provides the listener with a small amount of two dimensional discrimination but it is not the same as being there in person, when the acoustic ambience completely surrounds the listener.

Quadraphony

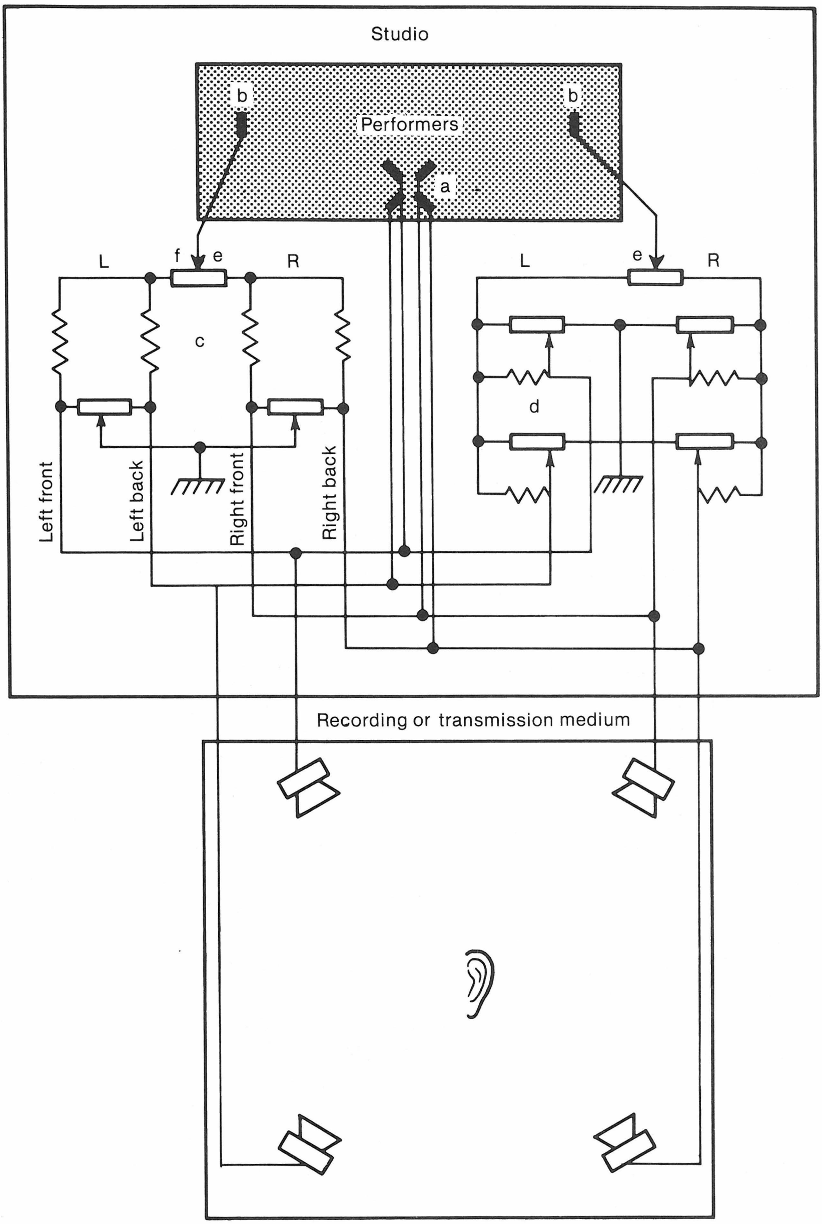
A closer approach to reality can be achieved by using four channels with four loudspeakers arranged to supply a sound image for each of the four diagonal quadrants around the listener. In its simplest terms it can consist of two stereophonic systems arranged back to back.

Practical considerations

Quadraphony can be most effective when used to reproduce the normal 'concert hall' situation, ie with the orchestra in front and the ambience to the sides and rear. Unfortunately the improvement is rather subtle when it comes to persuading the public to buy a four-channel amplifier and four loudspeakers with their attendant problems of accommodation. The temptation is, therefore, to make the effect more startling by 'panning' (p. 48) the sources so that they appear to be coming from all around the listener. Whether this effect is really desirable, with the possible exception of certain works specially written for spatial effect (eg Stockhausen's 'Gruppen'), is a matter of opinion but it is perhaps due to these sort of considerations that quadraphony has not made much impact to date.

Three-dimensional sound

Much of the advantage of quadraphony could be achieved with three channels. In fact it can be shown that all the horizontal information could be contained in three. The fourth channel could be used for the vertical dimension if there were no need to retain stereo compatability. As mentioned on page 16, although our hearing mechanism is designed to provide us with directional information mainly in the horizontal plane, we can also recognise the vertical element to a certain extent. This would make for realism but it would be a physically awkward format to accommodate in the average sitting room.



The performers can be grouped in front of or all around four microphones possibly arranged as two back-to-back stereo pairs (*a*). The outputs go eventually to four loudspeakers arranged on the four diagonal quadrants with respect to the listener. Additional 'spot' microphones (*b*) can be added to the balance and 'panned' in to the required apparent position. (*c*) and (*d*) are two types of front-to-back pan pots which can be fed from the stereo pan pots normally available in stereo consoles (*e*).

Using a system of 'matrixing' and four loudspeakers, it is possible to surround the listener with sound.

Surround Sound

The stereophonic effect

Conventional stereophony can provide a very effective aural image of a sound source but in only one plane. In the real-life situation, the listener in the concert hall would also be receiving reflected sounds from all around. In drama, dialogue and sound effects would come from all directions.

Ambisonic sound

The 'surround sound' effect can be simulated by matrixing (ie electronically combining in particular phase) two channels to provide outputs, containing the direct and ambient component, to supply additional loudspeakers. Decoders are available which generate signals for four or more loudspeakers in appropriate phase to suit their position. By combining three channels (or even two and a half channels because one need only have a bandwidth to 5 kHz) all three dimensions can be represented.

Simple ambisonic arrangement

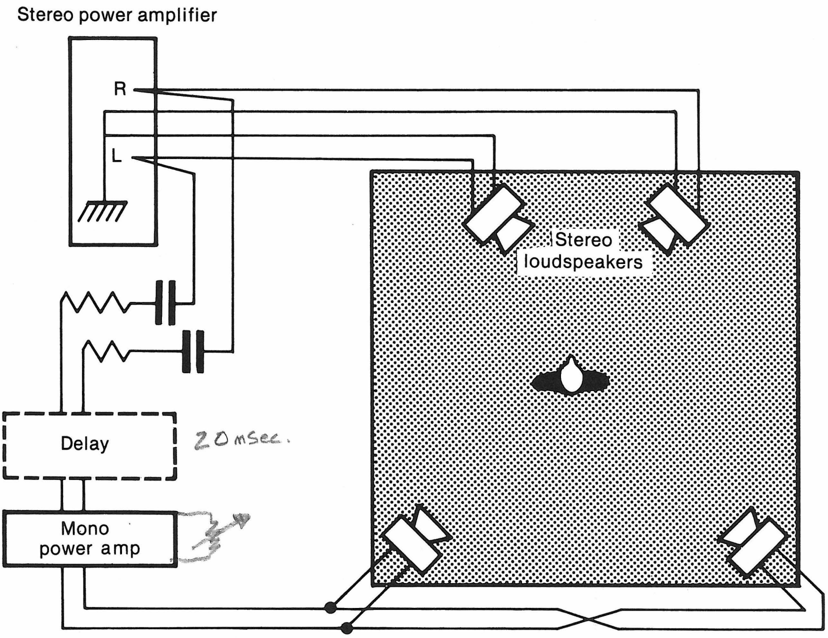
A simple method of obtaining an ambisonic effect from a stereophonic signal can be arranged as follows:

A stereo signal has two channels, left (L) and right (R), which are fed to the respective loudspeakers. Two other components can be derived from these two channels: the M signal, which is the sum of the two channels (L+R), and the S signal, which is the difference between them (L−R). The M (or mono) signal represents the central image of the sound and is normally compatible with monophonic reproduction. The S (or stereo) signal is composed of the off-centre element. In other words when an artist is performing in the middle of a 'sound stage', all the direct sound from the artist would be in the M signal. It would therefore be a simple matter to extract the S signal (eg by adding the L and R signals in antiphase) and feeding it to additional loudspeakers behind the listener. If the rear loudspeakers are themselves in antiphase with respect to the listener, the ambience appears to spread around the room in a very satisfactory manner.

The above technique is very simple and can work quite well particularly in association with sounds to which a straightforward microphone technique has been applied. It tends to widen the 'sound stage' in circumstances where multi-microphone balances are used in which the entire output of a source is only reproduced from one side or the other.

Use of delay

The realism of the ambient (S) channel can be enhanced by introducing a delay of about 20 msec into the output, thereby exploiting the *Haas effect* (see p. 28).



A simple method of obtaining a form of surround sound. Most stereo power amplifiers have one leg of each loudspeaker connection common to earth. Connecting between the two live terminals therefore produces an antiphase output (R–L). This can be applied to a mono amplifier and the output feed, preferably through a delay of about 20 msec, to the rear loudspeakers connected in antiphase, using the system to reproduce concert hall ambience. The rear loudspeakers do not require an extended bass response as this does not influence the stereophonic effect in a small room. Neither is extreme treble necessary as the more pleasant concert hall acoustics tend to favour mid frequency reverberation. Good quality, mid range loudspeaker units mounted at picture-rail level are quite satisfactory for this purpose.

Co-sited microphones for stereophony and quadraphony should have their elements as nearly coincident as possible.

Microphones for Stereophony and Surround Sound

The basis for many stereo, quadraphonic or surround sound balances is a number of coincident microphones aiming in different directions. In general, in this application, the nearer actual coincidence these microphones can be the better, otherwise the spacing between them can result in phase differences leading to cancellation in the higher frequencies.

The AKG C 422 stereo microphone
The AKG stereo microphone consists of two capacitor microphone capsules with a switchable polar response that can be varied between figure-of-eight and cardioid. The capsules are mounted one above the other in the same housing but one of the units can be rotated with respect to the other to adjust the angle of orientation between them.

The Calrec Soundfield microphone
The Calrec Soundfield microphone consists of four capacitor microphone capsules arranged in the form of a tetrahedron close together in a single casing. The outputs of these units are amplified and fed to a processor which combines them in certain proportions and phase relationships according to a complex mathematical formula to vary the effective directionality of the microphone.

The four outputs can be supplied for direct recording representing the three orthogonal components of pressure gradient (ie phase difference between front and back): left-minus-right, front-minus-back and up-minus-down.

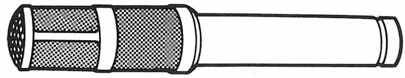
Alternatively these outputs can be processed to provide any type of directional characteristic, from omni-directional through cardioid and hypercardioid to figure-of-eight.

The signals can be further processed to provide a variety of outputs and directional responses.

These could consist of a single monophonic output or a stereo pair. In each case, as well as providing adjustable polar characteristics, the effective direction of the microphones can be 'panned' or tilted in any direction. There is also a dominance control to vary the vertical or forward emphasis for optimum balance between incident and ambient sound. There is also a four-output mode which effectively provides four microphones pointing in four directions with adjustable angle for quadraphonic recording and ambisonic matrixing for surround sound recording reproduction.

The ability to make all these adjustments of apparent orientation electronically and remotely can be a considerable asset in the recording of concert performances.

1, The AKG C 422 stereo microphone. The angle between the two capsules can be varied by rotating the head.



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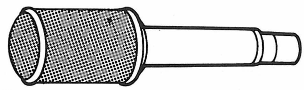

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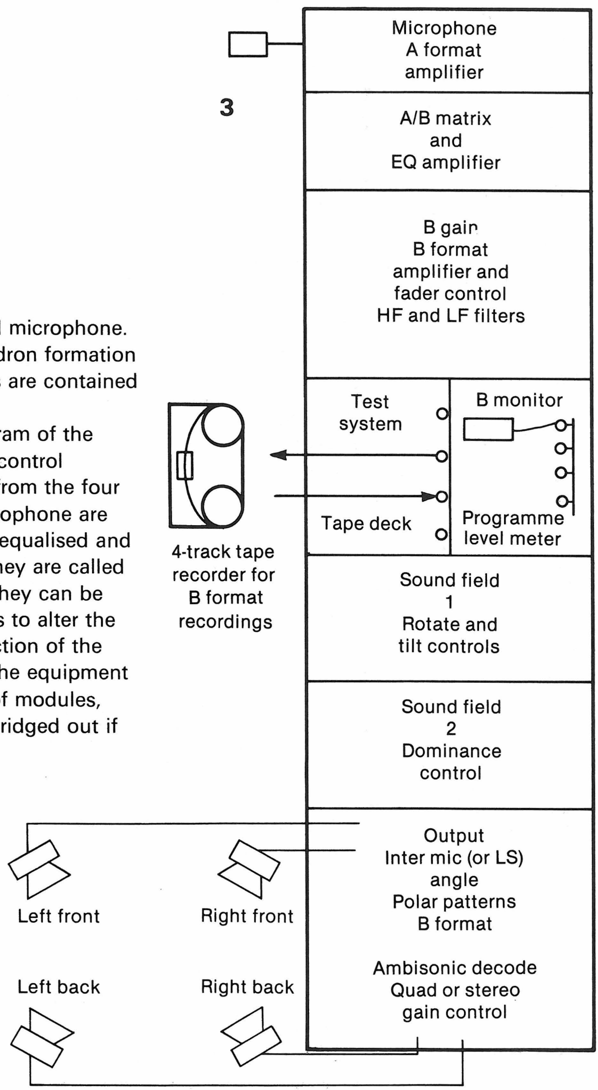


2, The Calrec Soundfield microphone. Four capsules in tetrahedron formation and their head amplifiers are contained in a single housing.

3, Block schematic diagram of the Soundfield microphone control equipment. The signals from the four capsules within the microphone are called 'A format'. When equalised and converted to line level they are called 'B format'. In this form they can be matrixed in various ways to alter the effective angle and direction of the microphone response. The equipment is supplied in the form of modules, some of which can be bridged out if not required.

Frequency Response Control

Frequency response control, or 'equalisation' as it is usually called, plays an important part in present-day sound technique.

Response selection amplifiers
Most professional sound recording consoles provide response selection amplifiers in each of the input channels. These normally provide up to about 15 dB of available high or low frequency lift or cut. Some also provide 'presence' lifts or cuts, ie a peak or trough centred on a choice of frequencies in the region of 3–5 kHz. This is the region concerned with the sibilant frequencies in speech which give the effect of closeness.

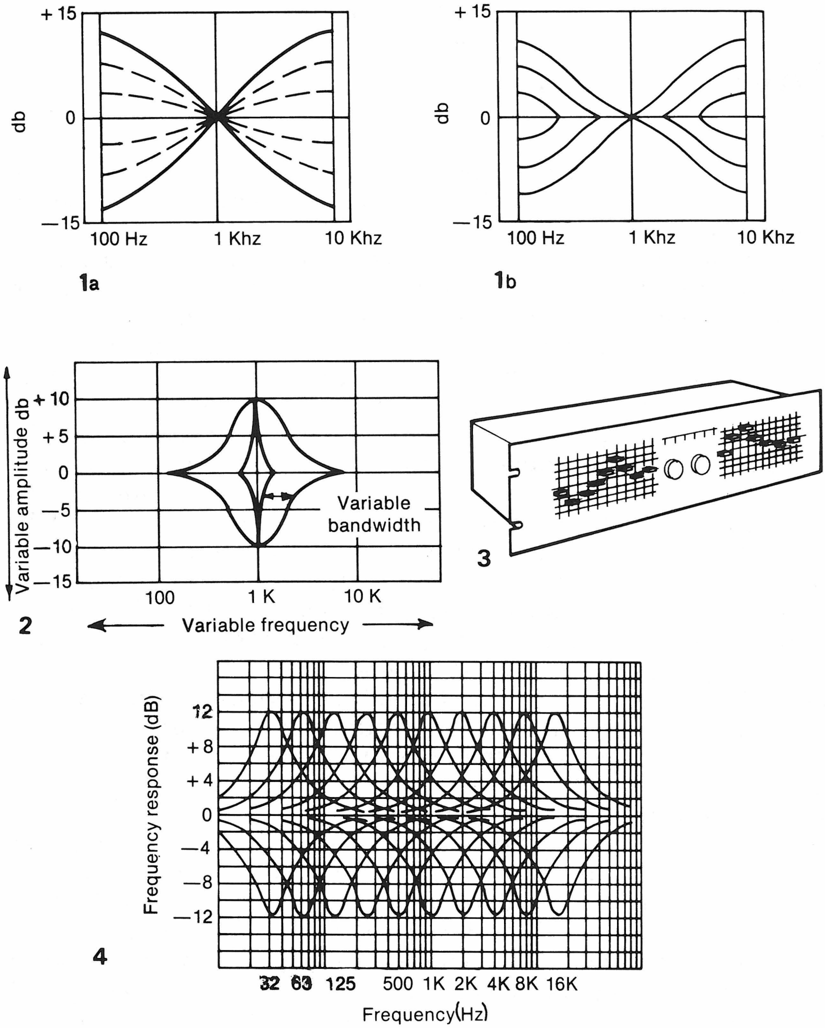
Parametric equalisers
Parametric (or variable parameter) equalisers are a useful tool in the recording studio or listening room. They provide a variable peak or trough of up to about 14 dB. The frequency and bandwidth can be varied throughout the audio spectrum.

With a well designed parametric equaliser, it is possible to apply frequency response adjustment very precisely and selectively, for example to correct for a response peak due to acoustic resonance in the studio or listening room. Alternatively, the equaliser can be used for artistic effect, for example to give 'presence' to a voice or greater impact to an instrument.

Graphic equalisers
Graphic equalisers are a very sophisticated form of tone control in which the audio spectrum is divided into narrow bands each centred on a specific frequency which is usually based on either octave or third-octave intervals. Each has an individual slider control, giving an increase or decrease of up to 15 dB.

The term 'graphic' stems from the fact that the positions of the sliders give a graphic indication of the shape of the response curve selected. They provide an excellent method of equalising the frequency response of equipment or compensating for acoustic coloration. They can be particularly useful for matching equipment, such as microphones and loudspeakers.

Bypass switches
One problem associated with the use of frequency response adjustment is that it is all too easy to become accustomed to a particular quality of sound. Most equalisers are therefore provided with bypass switches by which the equalisation can be defeated to make comparison with the straight-through condition.



1, *Response selection amplifiers. a.* In one type the bass and treble lift/cut is adjusted by varying the slope of the response curves about a fixed middle point, in this case 1 kHz. *b.* In another type the bass and treble lift/cut is altered by varying the turnover frequencies.

2, *Parametric equalisers.* The bandwidth of the peak or trough can be adjusted between about 0.1 and 5 octaves. Its amplitude can be adjusted between zero and about ±15dB and its position slid along the frequency scale.

3, *A typical stereo graphic equaliser.* This type of equaliser gives control to ±12dB over 10 octaves centred on 32, 63, 125, 250, 500, 1k, 2k, 4k, 8k and 16kHz. The positions of the sliders give a 'graphic' indication of the shape of the response curve.

4, *Graphic equaliser curves.* Careful design is necessary to prevent phasing distortion due to interaction between the overlapping peaks. Most graphic equalisers produce a response that looks very distorted when viewed on an oscilloscope but with good design this is not audible.

Sound control is necessary for technical and artistic reasons.

Sound Control

Practically every type of recorded, amplified or broadcast material requires its volume controlled. This is necessary for a variety of reasons, the most important of which are:

1. To keep within the dynamic range of the system. In most cases the technical upper limit is set by the onset of overload distortion and the lower limit by the threshold of intrusive noise.
2. To limit the volume range of the programme material to suit domestic listening conditions. Most listeners listen to programmes at a much lower volume level than in real life, particularly in the case of music and dramatic productions. They tend to set the volume so that the loudest parts of the programme are at a comfortable level. If no compression were applied to the volume range the quieter passages would either be inaudible or submerged in the general domestic background noise.
3. Control is necessary for artistic reasons to select and balance between the various elements in the sound source. Even where only one source is concerned, volume control is used to direct emphasis by relating volume to importance.
4. To ensure that programmes recorded or broadcast at different times match each other and can be reproduced successively without the necessity for the listener to adjust his volume control.

Multi-microphone technique

Sound control increases in complexity as the number of microphones increases. When two or more are employed, it is necessary not only to keep the level of each within the technical limits but also to balance the outputs for artistic effect. The whole essence of artistic control is that it is exercised where it is required and not only on the combined output. Artistic control can be done simultaneously or, in the case of multi-track recording, in two separate operations, ie each output is recorded to the optimum level within the technical parameters on separate tracks and then, at a later stage, the tracks 'mixed-down' and dubbed to two for the final stereo output.

Use of loudspeakers

Effective programme control requires the use of meters to measure the volume level for technical purposes but no meter can make artistic judgements or even give an accurate assessment of loudness, which is largely subjective.

Effective programme control can only be obtained by means of a good quality loudspeaker, using the meter to 'calibrate' the ear, much as a car driver judges his speed with only occasional glimpses at the speedometer.

Speech modulation

1, Most sources of sound are continually varying and interrupted in character and a fade which continues through the gaps in the material would result in a succession of steps in the output.

Unobtrusive control

2, If the intention is for the operation to be as inconspicuous as possible, which is the case for most types of material and particularly music, the action should occur in the gaps in the material or during rapid fluctuation of level.

Control for dramatic effect

3, If it is intended to make an obvious smooth fade for dramatic effect (eg. a slow fade on dialogue is the conventional method of suggesting a change of scene in radio drama) movement of the control must occur only during the material, preferably during sustained phrases.

Compressors enable higher average sound levels to be maintained without overload.

Automatic Control

No automatic device can provide the necessary aesthetic judgement and finesse for effective programme sound control. Nevertheless automatic control systems (known as compressors and limiters) have considerable application in sound operation practice to assist in the 'taming' of wildly varying and unpredictable sources and to protect equipment from possible overmodulation.

The limiter

The limiter is a device through which programmes can be passed without alteration of the signal until a critical value is reached. If the input signal rises above this value (usually called the onset point), the gain of the system is automatically reduced (below unity) so that the output cannot rise significantly above the limiting value. This limiting action is caused by reduction of the amplifier gain not merely by cutting off the peaks of the waveform (peak-chopping results in very severe distortion).

The compressor

The compressor is similar to the limiter, in that above the onset point the gain of the system is reduced, but its action is less dramatic so that an increase in input level above this value produces a reduced increase in the output. The extent of the gain reduction is usually adjustable and is called the compression ratio. Other controls usually available on compressors are:

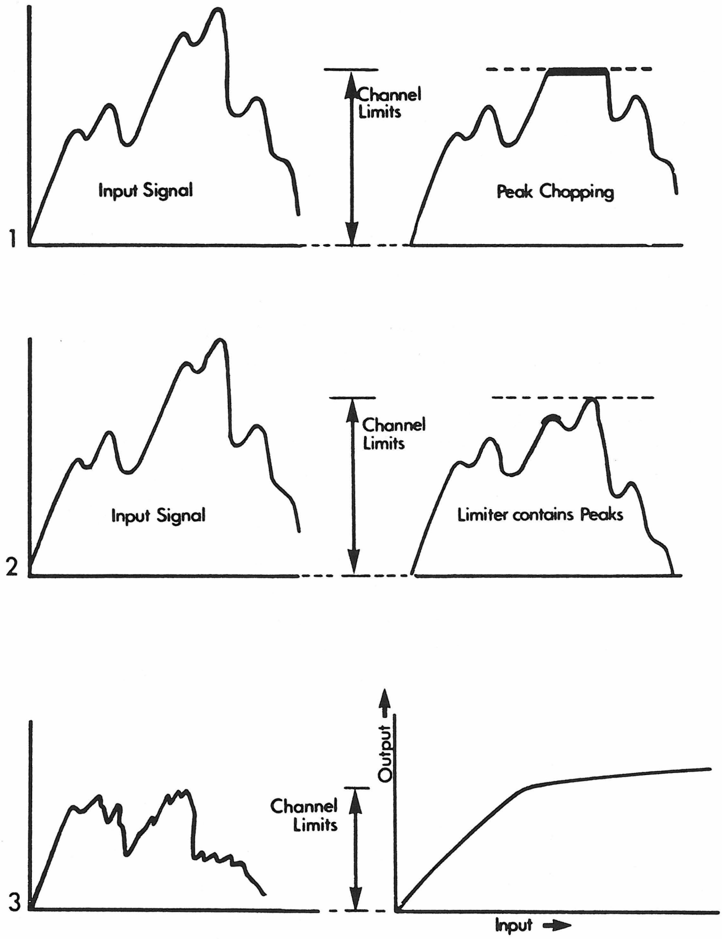
1. The onset or threshold point (the value at which the action commences).
2. The attack time, ie the time required for the action to take effect following a sudden increase in the level.
3. The recovery time, which is approximately the time required for the gain to be restored to unity after the high level signal has been removed.

The attack time is set as a compromise between too fast, which can alter the shape of the waveform and thereby cause distortion, and too slow with consequent overshoot.

Recovery time is usually set to make the variations of gain as inconspicuous as possible without allowing isolated peaks to depress the level for too long.

The noise gate

A noise gate is almost the opposite of a compressor. It reduces the amplification below a certain threshold level (thereby reducing noise in the absence of signal) and increases it when the minimum acceptable programme level is reached. It thus acts as an 'expander' to increase the distinction between signal and noise level.



AUTOMATIC CONTROL

Compressors and limiters

1, Effect of peak chopping with severe distortion. 2, A limiter should cause gain reduction to contain the peaks of the waveform. 3, Input/output curve for a limiter showing the 'knee' or onset point of compression.

The search for perfection leads to automated sound mixing.

Automated Sound Mixing

Recording technique has advanced to the stage where, at least in the 'pop' music field, the process has become a part of the performance.

Multi-tracking technique

The complexity of modern balance technique is such that it is almost impossible to obtain the best results by mixing the sounds as they are made. It is much better to record each element on separate tracks of a multitrack recorder so that each can be individually treated and, if necessary, repeated without affecting the rest. Moreover most of the treatment and manipulation of the sound can go on after the musicians have left the studio. However, this 'mix down' process can be tedious and needs to be efficiently handled if optimum results are to be obtained without extravagant use of studio time and this has led led to the introduction of *automated mix-down*.

Automated mix-down

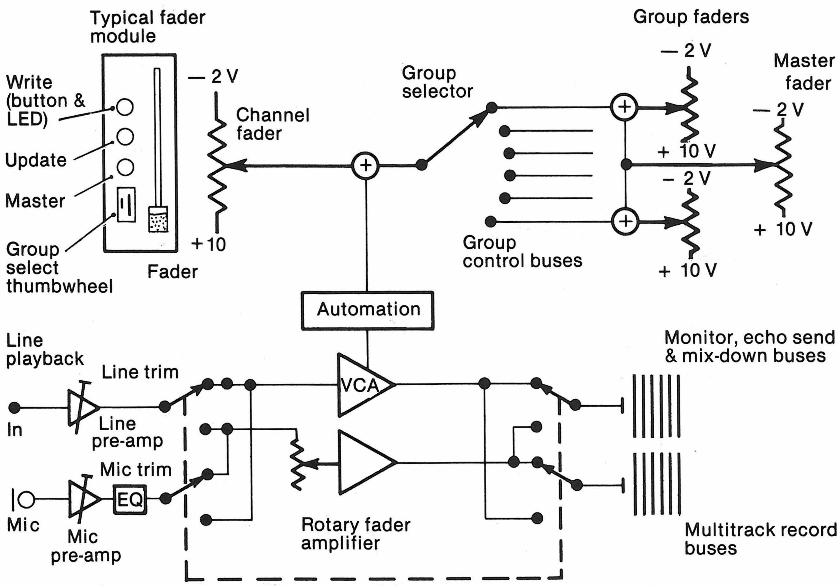
During the first attempt at mix-down a signal is recorded on one track of the multi-track tape which, in one system, is coded to represent the variations of the individual channel faders in relation to position along the tape. Alternatively a time code (usually the SMPTE time code) can be recorded and used to synchronise a 'floppy disc' memory system for the same purpose. Once the fader settings have been memorised on the tape, the mix can be repeated and the channel faders modified as required (and the memory updated) until an optimum mix is obtained for the master two-track or four-track tape recording.

Mechanical fader control

The method by which the fader settings are reproduced can be mechanical or electronic. In the mechanical system the faders are physically moved by servo motors and can be seen to be operating as though by an invisible human hand. Touching the fader knob has the effect of disabling the fader servo so that the settings can be modified and re-memorised by this natural and instinctive operation.

Electronic channel control

In an alternative method, the effect of the faders and not their position is automated. This technique is facilitated by the use of voltage-controlled amplifiers (VCA). These amplifiers have their gain controlled by a DC voltage, typically in the range -2 V to $+10$ V DC. The control voltage adjusts the amplifier gain logarithmically and is usually arranged so that an increase of one volt represents an attenuation of 10 dB. VCAs are usually set up so that zero volts from the control fader produces zero gain, $+10$ V gives 100 dB attenuation (cut off) and -2 V gives 20 dB gain. This allows for 20 dB 'headroom' above normal settings.



The control voltage for the VCAs is varied by the channel faders. It can also be supplied from group faders by selecting the appropriate bus bar to add to the fader control voltage.

Finally the group faders can be controlled by master faders by simply adding their voltage to the groups, the actual volume adjustment still taking place in the channel faders.

One advantage of this system is that the individual channels can be controlled as groups without their outputs being mixed so that they can be recorded completely separately on the multi-track tape.

To make changes in the mix on subsequent attempts it is necessary to find the position on the fader that matches the VCA control voltage. This is shown by two LEDs one of which lights up when the fader is too high and the other when it is too low. When both light, the fader setting is matched; the 'write' button can be pressed and the fader moved to a new position to update the memory.

Some mixing consoles have alternative paths (controlled by rotary faders) for making the original recording.

Volume metering is essential for good recording to maintain levels between the parameters system noise and overload.

Programme Meters

Twin lamp systems

Some simple recorders employ two lamps (usually light emitting diodes). One lights up to show that minimum recording level has been achieved and the other when the overload point is reached.

Volume unit meter

The majority of tape recorders have volume unit meters to indicate volume. In the simplest form, these can consist of a simple moving-coil meter and rectifier which gives an indication of signal level the accuracy of which varies with the type of programme being measured. All meters can be made to give an accurate indication of zero level (OVU) on steady tone but the way they react to programme material depends upon the ballistics of the meter (rise time and overshoot) and the value of the resistance through which it is connected. Meters used for professional purposes, where levels have to be set and compared, should conform to the American Standards Association specification. Even these will give widely different, but matching, readings with different material. Due to the slow rise time of the meter, impulsive sounds, such as speech, starting from a low level will read lower than their actual volume level; sustained sounds, ie long notes, will read higher and fluctuations near the top of the scale can overshoot and read high.

For this reason VU meters have to be interpreted from experience rather than just read; line up level is usually taken to be +4 dBU (1.73 volts) to read 0 VU to give an actual peak recording level of +8 dBU on average material.

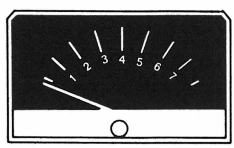
The peak programme meter

A much more accurate visual indication of volume level is the peak programme meter. The PPM has a rapid rise time (2.5 msec) and a slow recovery time (approx 3 sec) actuated by charging a capacitor and discharging it through a high resistance.

This action makes the meter easy to read and it gives a reasonably accurate indication of peaks which is the most important consideration.

Visual displays

The advent of multi-track recording has brought the need for large numbers of meters to be displayed simultaneously. The most convenient way of arranging this is to have indicators consisting of a stack of LEDs which light up in sequence in relation to volume level. The circuitry can follow the PPM principle which helps to reduce rapid flicker.



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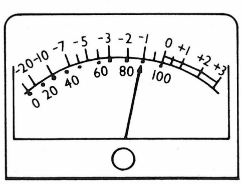

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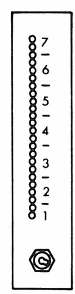


1, Peak programme meter. The figures are white on black to reduce eye strain. The low is quasi-logarithmic so that the scale is reasonably evenly spaced, with 4 dB between each of figures 2–7. Line up level is usually set to 4 and maximum level (+8 dBm) is 6. The extra 4 dB to 7 dB is to give an indication of the degree of overload; 2, The VU meter. There are two scales. The upper one is marked in decibels (red above the zero mark). It is only accurate on standby tone. The zero mark represents full modulation and the figures in the upper (red) section the degree of overload. The lower scale is marked 0–100 and represents the percentage of full modulation. The scale is uneven, being largely cramped towards the top end. The rapid fluctuations make it difficult to read and especially to compare readings; 3, A visual display plate level meter using a column of light-emitting diodes. Different colours can be used to represent low, normal and overload levels.

Dolby A is an intermediary process for reducing noise in sound recording.

Dynamic Noise Reduction— The Dolby A System

Modern recording techniques usually involve the re-recording (dubbing) of programme material from one medium to another, in some cases many times over. This can be for the purpose of 'reducing' a multi-track recording to twin track stereo, for editing or for transfer to disc etc. Each recording will add its quota of noise to the final result.

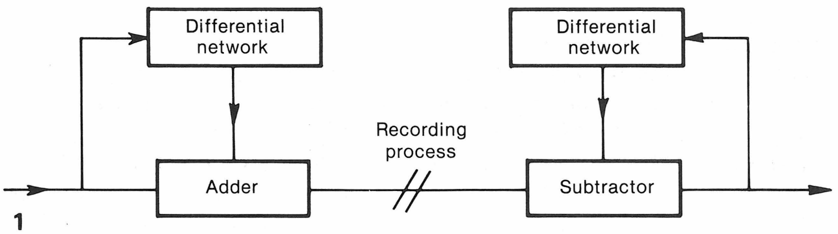
Companding

In order to improve the signal: noise ratio a system of 'companding' can be used, ie the volume range of the signal is compressed to raise the average level before recording and then expanded, depressing the quieter elements of the programme (and with it the noise) on reproduction.

Unfortunately there are two main problems in designing a companding system that does not cause more trouble than it cures. One is the difficulty of making the compression and expansion systems 'track' perfectly over the whole frequency and volume range. The other is preventing the background noise level going up and down with the signal, which only makes it more obvious. *The Dolby A system* overcomes this problem in the following ways.

The signal is provided with two parallel paths, one via a linear amplifier and the other via a differential network, the output of which is added to the 'straight through' signal when recording and subtracted when reproducing. The differential network divides the frequency spectrum into four bands and treats each one separately so that the action is applied only where it is needed. With music for instance most of the sound energy tends to be concentrated in the lower middle band whereas tape hiss is largely high and too far removed in frequency for 'masking' to take effect. This is the effect whereby the ear tends to be deaf to sounds in the presence of, and for a short time after, hearing louder sounds of similar frequency. The Dolby system exploits this effect by restricting the action to narrow frequency bands.

The system is adjusted so that with a low level input (below −40 dB) the output is 10 dB higher than the input for frequencies up to about 5 kHz above which it increases progressively to 15 dB gain at 15 kHz. As the input signal rises above +40 dB, it is progressively less affected. At high levels (where the possible ill effects of companding are most obvious), the compander has least effect and is virtually bypassed. The Dolby 'A' system is normally used only as an intermediary process, the final product having a straightforward response. This is in contrast to the Dolby 'B' process where the record is sold with the modified characteristic incorporated on the assumption that the complimentary characteristic will be applied in the reproducing apparatus.



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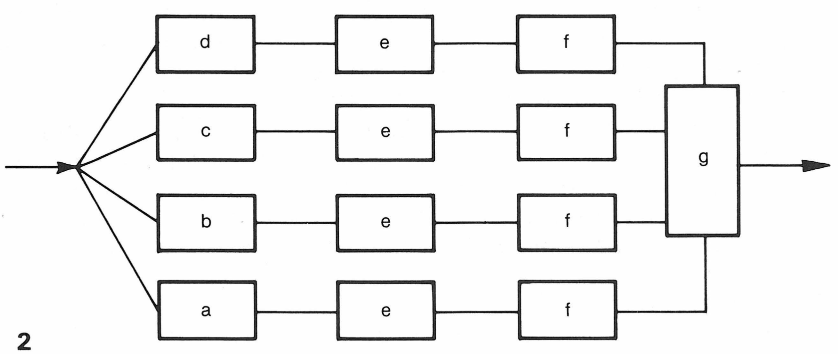

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1, Companding the recording process. The output of the differential network is added for recording and then subtracted when replaying; 2, Schematic detail of differential network: *a* 80 Hz low pass filter to deal with hum and rumble; *b* 80–3000 Hz band pass filter to deal with mid band noise and print through; *c*: 3000 Hz high pass filter to suppress hiss and modulation noise; *d*: 9000 Hz high pass filter to suppress hiss and modulation noise; *e*: linear limiters; *f*: non-linear limiters; *g*: adder.

The Dolby B noise reduction system is used in high quality domestic cassette recorders to compensate for tape hiss due to slow speed and narrow tracks.

The Dolby B Noise Reduction System

A simpler system than the Dolby A process is the Dolby B system which is used mainly in the domestic tape cassette market to reduce the high background level of tape hiss which is inherent in this type of recording due to the low tape speed and narrow track. With Dolby B the tapes are recorded and marketed with the Dolby characteristic on the assumption that they will be reproduced by equipment with the complementary characteristic.

The Dolby B system

As in the Dolby A system, there is a main programme path and a side chain. The side chain incorporates a compressor which is preceded by a high pass variable filter covering frequencies from about 500 Hz upwards. In the record mode, signals below a given threshold level are boosted by the compressor and added to the side chain. This increase in level is applied progressively by the variable filter from 500 Hz up to 10 dB at 10 kHz.

Thus low level high frequency signals are recorded up to 10 dB higher than the original. An overshoot suppressor (diode clipper) is provided, following the compressor, to prevent high level transients, which are faster than the time constant of the compressor, being added to the output. Any resultant distortion is masked by the high level signal and tends to cancel on replay.

The Dolby B decoder, which is fitted to most high quality cassette machines, is identical to the coder except that the side chain is fed with a phase-inverted signal so that the output of the compressor is added to the main chain in antiphase and is effectively subtracted from it thereby producing an exact complement of the coding process.

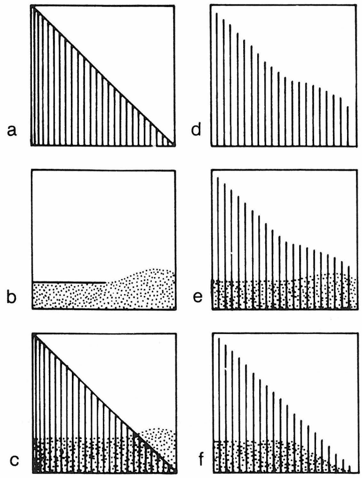
Noise reduction

As the low-level high frequency response is reduced on playback so is the tape hiss and system noise that has been introduced by the recording, giving an improvement in signal: noise ratio of up to 10 dB.

The difference between the Dolby system and the simple application of pre-emphasis to the recording and de-emphasis on playback (which is applied anyway) is that the Dolby B characteristic only affects low level signals.

Compatibility

Dolby B encoded material can be played on equipment without the Dolby function if the high frequency response is reduced to compensate for the Dolby characteristic but this does result in a lack of high frequency in the louder passages of the music.



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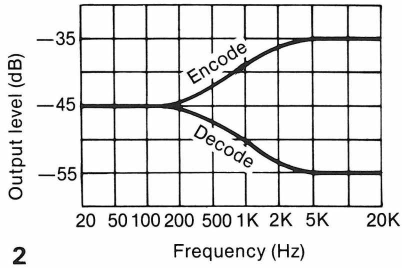

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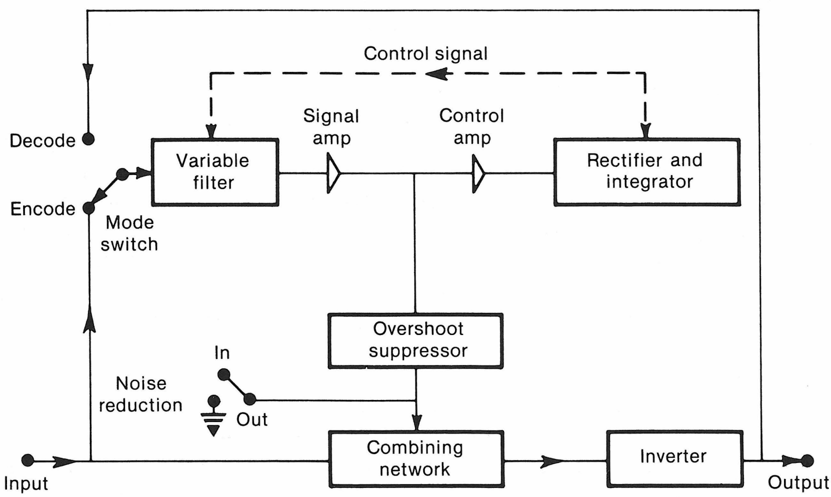

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1, *a* A representation of the distribution of energy in average programme material. The energy tends to decrease as frequency increases. *b* Tape noise. *c* In normal reproduction low-level high-frequency signals are masked by tape hiss. *d* Dolby B encoded signal with compression applied to boost low-level high frequency signals. *e* When the tape noise is added the boosted high frequency signal is still above the noise level. *f* Decoding reduces the low-level high frequency response to normal and with it the noise; 2, Complementary response curves of a Dolby processor with low-level input; 3, Block schematic diagram of a Dolby B encode/decoder.

The DBX and DNL noise suppression systems rely on the premise that high frequency energy tends to increase disproportionately with a general increase in volume.

The DBX and DNL Noise Suppression Systems

DBX noise suppression, like Dolby A, is a complimentary system.

DBX

The DBX system employs a wide range, 2:1, compression ratio to encode and decode. Problems of tracking the compression and expansion are eased by the straightforward ratio and the fact that the level sensing is done on an RMS basis, ie controlled by the total power of the signal, regardless of the phase relationships of the various components.

The DBX system exploits the fact that, in most programme material, the bulk of the power is in the low frequencies and that high power in the high frequencies occurs only when the general volume is large.

Pre-emphasised compression

The signal fed to the compressor is heavily pre-emphasised to increase the overall power on recording. It is similarly de-emphasised to restore it to normal, while at the same time reducing the high frequency noise, when it is decoded. To prevent the recording being overloaded by any powerful pre-emphasised high frequency signals, a similar pre-emphasis is applied to the side chain of the compressor so that, in effect, the high frequency recording level increases as the frequency increases and as the high frequency level decreases.

Masking effect

DBX also takes into account the masking effect of our ears which makes us relatively insensitive to noise (in this case high frequency noise), in the presence of loud sound of a similar frequency. The use of DBX can result in an improvement in high frequency signal: noise ratio of up to 30 dB.

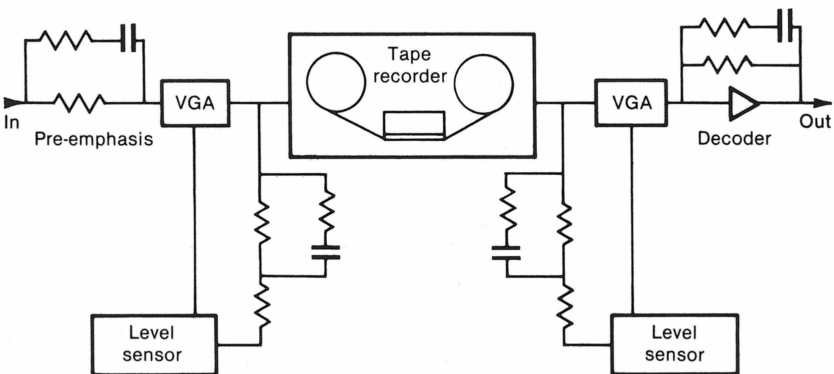
The Philips DNL system

The Philips DNL system operates on replay only and, therefore, must reduce the fidelity of the recording to some extent. As most musical instruments have fundamental frequencies below 4.5 kHz and when played softly produce few harmonics, it is considered that a high pass filter operating above this frequency has little effect on quiet material.

The system is basically a dynamic treble cut filter operating above 4.5 kHz on low amplitude signals.

When the signal exceeds the predetermined level, the filter is bypassed thus leaving the harmonics unaffected.

The system is claimed to give an improvement in signal: noise level of 10 dB at 6 kHz, 20 dB at 10 kHz.



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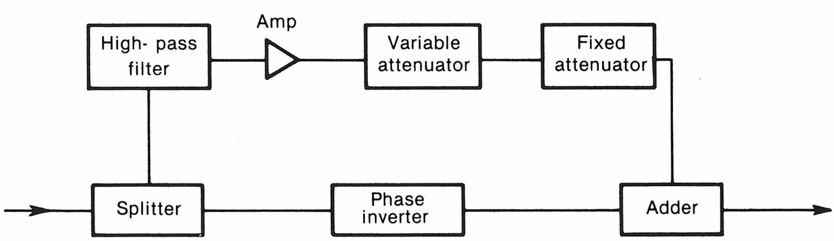

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1, Block diagram of the DBX noise reduction system. The variable gain amplifiers have a compression/expansion ratio of 2:1. Due to pre-emphasis, the increase is applied progressively to the high frequencies; 2, Block diagram of the Philips Dynamic Noise Limiter. This works on replay only and reduces the high frequency response when the high frequency component of the signal is low.

Basic Magnetic Recording System

Practically all recording nowadays, even when the final product is a disc, begins with the use of magnetic tape. This is because of its convenience and flexibility, particularly from the point of view of editing. Magnetic tape can be played back immediately after it has been recorded and, in the case of professional machines, monitored as it is being recorded. It is possible to erase and re-use the tape; several tracks can be recorded simultaneously and in synchronism and, above all, magnetic tape is capable of producing recordings of the highest standards.

The basic recording system
Tape recorders are made in a wide variety of types and standards of complexity and quality but the basic system consists in essence of the following:

1. A recording amplifier which processes the signal prior to recording.
2. A recording transducer (tape head) which translates the electrical signals into variations of magnetic force.
3. A storage medium, ie the tape.
4. A transport system to move the tape past the head at a steady speed so as to equate the variations of magnetic force with time.
5. A reproducing transducer (the original recording or a separate reproducing head) to convert the stored magnetic variations back to electric signals.
6. A reproducing amplifer to process the signal from the reproducing head to provide the output.

Tape transport
The standard direction of travel of the tape is from left to right as viewed from opposite the front of the tape heads. In some machines it is possible to reverse the direction of the tape in order to allow for very long record/replay times by playing alternate tracks (narrow bands along the length of the tape on which the recording is made) in opposite directions.

Mechanical systems
The tape is pulled past the heads by means of a capstan spindle against which it is pressed by a rubber-covered pinch roller. In some machines the tape storage and take-up spools are driven by the same electric motor as the capstan through a system of pulleys. In high quality machines separate motors are used for each function.

Most recording machines can erase previously recorded material from the tape prior to recording. This is achieved by means of an *erase* head over which the tape passes before reaching the recording head. There is also provision for rapid spooling of the tape in either direction to enable it to be rewound or wound on to cue.

A basic record/replay system. A single motor drives the capstan drive spindle and the storage and take-up spools through a system of pulleys.

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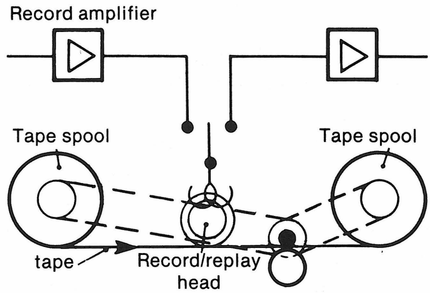

A domestic machine incorporating an erase head (*e*) but with a shared record/replay head. Tape movement and spooling is controlled by a single motor. The tape is held against the head by a pressure pad.

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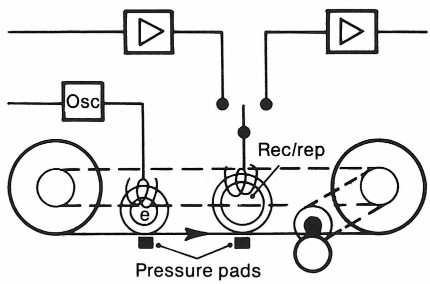

High-quality machines employ separate record and replay heads. Separate motors are used to drive the tape capstan spindle and each of the tape spools. This allows for proper control of tension of the tape against the heads without the aid of pressure pads.

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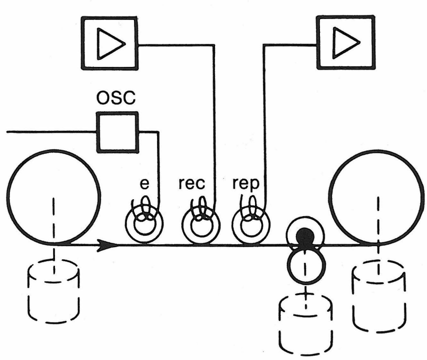


Magnetism is a basic force that occurs as a natural phenomenon in certain minerals.

Magnetism

Magnetic recording is a method of storing information in a material in terms of its state of magnetism.

Permanent magnets

Many hundreds of years ago it was discovered that pieces of natural iron ore called 'lodestones' were found to attract other pieces of iron and to attract or repel other lodestones according to certain laws. Moreover, if they were freely suspended by means of a thread they would turn in a particular direction in relation to the north/south poles of the earth. The material, magnetite, which is an oxide of iron (Fe_3O_4) was particularly pronounced in minerals located in the region of Magnetia, a city in Thessaly—hence the name 'magnet'.

Magnetic poles

Magnetism is found to increase towards the ends of a magnet, the longer and thinner it is the more this is the case. The two ends, called the poles, are nominated north (seeking) and south (seeking) according to which of the earth's magnetic poles they point to when freely suspended.

Magnetic field

If a compass needle is moved in the area of influence of a magnet (known as its 'magnetic field') and the direction it indicates is plotted, the needle will be found to trace out a series of concentric circles in the directions followed by what are called *lines of magnetic force*. The direction of these lines of force in a magnetic field is taken, by convention, to pass through the magnet in the direction from south to north and outside the magnet from north to south.

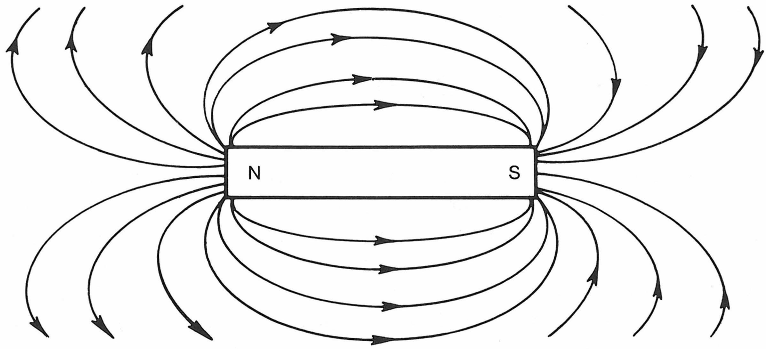
Magnetic field strength

The total number of lines of force that form the magnetic field is called the *magnetic flux* (symbol Φ). These lines tend to be unevenly distributed so that the degree of magnetic effect at a particular point depends upon the concentration of flux or *flux density* in that area. Magnetic flux density (symbol B) is proportional to $\Phi \div$ area.

Magnetic effect

If the opposite poles of two magnets are brought together lines of force will link, forming a bond and they will attract each other.

If like poles are brought together the lines of force, being unable to cross because they represent the resultant of the forces existing at the poles compress, causing repulsion. The force of attraction or repulsion between two magnets is inversely proportional to the distance between them.



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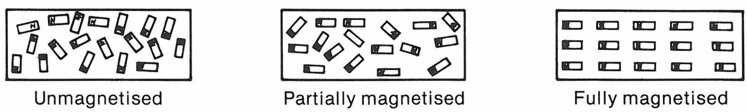

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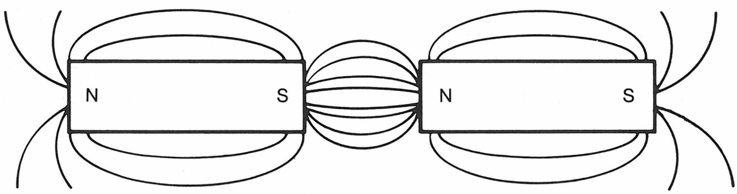

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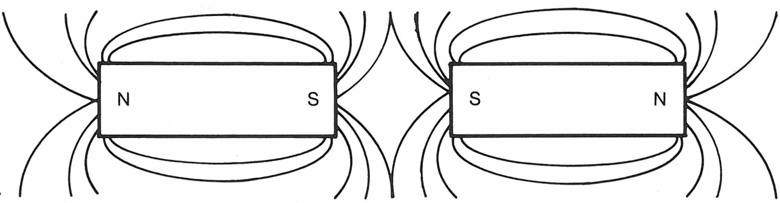

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1, Illustrating the lines of force in a bar magnet. The existence of these lines can be demonstrated by placing a card sprinkled with iron filings over the magnet and tapping it lightly. The filings will form up in the manner shown, but of course the effect occurs in three dimensions; 2, Matter is composed of molecules each of which can be considered as a minute magnet. In an unmagnetised substance they are arranged in haphazard fashion and have no overall magnetic effect. In a partially magnetised substance, many of the molecules have their magnetism orientated in the same direction. In a fully magnetised (saturated) state virtually all the molecules are turned the same way and their magnetic effect adds up; 3, Lines of force tend to shorten themselves, ie contract along their length. This tends to draw opposite poles together; 4, Lines of force tend to expand laterally, ie resist lateral compression; as they cannot cross, like poles repel each other.

A changing magnetic field that cuts a conductor induces into it an emf proportional to the rate of change.

Magnetic Induction

When an electrical current flows in a conductor, lines of magnetic force are set up in and around the conductor. It can be shown that the movement of a conductor in a magnetic field, or the movement of a magnetic field which causes lines of force to cut the conductor, will induce an emf into the conductor and, if there is a circuit, cause a current to flow through it. Thus, a reversal of the process by which information in the form of electrical signals can be stored, by electrically magnetising an element, can be used to retrieve the information by causing the magnetised element to move in relation to the conductor or coil.

Mutual induction

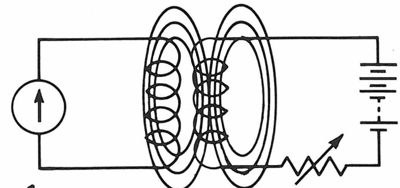
If two coils of wire are arranged in close proximity, so that the lines of force produced by one also pass through the other, and a battery is connected to one coil and a galvanometer (an instrument that indicates the strength and direction of an electrical current) to the other the following phenomena can be observed. When the battery is connected, the galvanometer will briefly indicate a high current flow and then return to zero. When the current is switched off, the galvanometer will indicate a brief current flow in the reverse direction. If the polarity of the battery is reversed, the same will occur but in the reverse sense. If the switch is exchanged for a variable resistance a similar reaction can be produced but only while the resistance is being altered; the extent of the galvanometer deflection then depends on the rate of change of the control. Again, if the resistance is altered in the reverse direction the galvanometer will indicate opposite deflection.

The value of the *electromagnetic induction* depends upon the rate of change of the flux linkages between the circuits. The most obvious practical example of the use of this phenomenon is the transformer, in which the two coils are wound on the same core to achieve maxi-maximum flux linkage. Instead of varying a direct current, an alternating current is used so that the magnetic field is continually building up and collapsing upon itself with the repeated changes in polarity.

Self induction

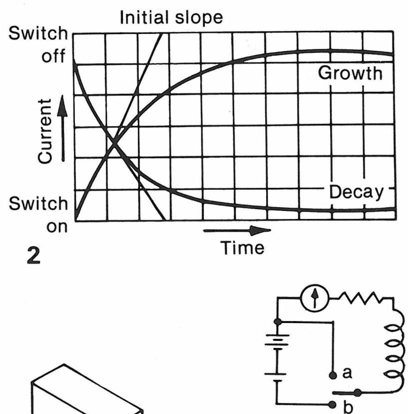
If a changing magnetic field around a conductor or coil can induce an emf into an adjacent conductor, it will obviously have the same effect upon itself. Changing the current flow through a conductor produces a change in the magnetic field surrounding it. In cutting the conductor, the change in the field induces an emf which tends to maintain the status quo, ie it tends to increase a decreasing current flow or vice versa. This property of induction, which is analogous to inertia in mechanical terms (in that it tends to resist change), is proportional to frequency, ie rate of change.

1, In this circuit while the resistance is being adjusted the galvanometer will deflect, showing that the current in the secondary coil is endeavouring to oppose any state of change in the flux linking the two coils.



1

2, *Time constant.* When a circuit has inductance, the action of switching a current on or off converts electrical energy to magnetic energy or vice versa. This takes a finite time.
In the circuit, if the switch is made to B, were it not for the inductance, when this circuit was switched on, the current would rise instantly to the full value determined by the total resistance of the circuit.
Due to the transfer of energy caused by the inductance, the current rises more slowly (the growth curve).
The recriprocal situation occurs if the inductance is short-circuited by the switch. The collapse of the magnetic field induces a current to flow. The time that the current takes to rise to 63.2% of its final steady value is the *time constant* (which equals inductance+resistance).



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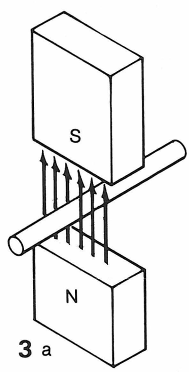
3, The effect of passing current through the conductor is to cause an increase in the flux on one side and a decrease on the other, giving rise to a mechanical force (F) in the direction indicated.



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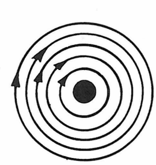

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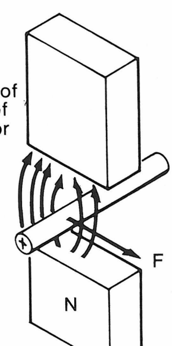
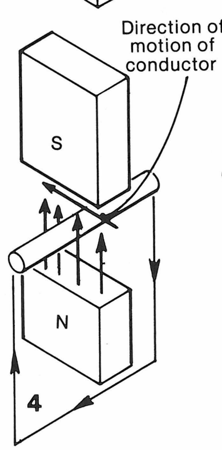


3 c

4, Induction through movement in a magnetic field. If the conductor is moved at right angles to a magnetic field, a current will be induced in the direction that produces a force which tends to oppose the motion (see above).



4

When a ferrous material is magnetised alternately in opposite polarity the magnetising characteristic does not retrace itself but forms into a 'hysteresis' loop

Magnetisation Characteristic

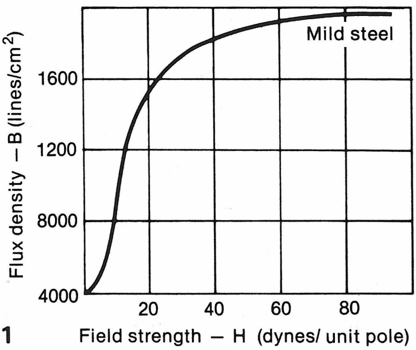
On the preceding page it was shown that a piece of ferromagnetic material can be magnetised by the passage of an electrical current through a coil of wire surrounding it. The extent to which the material remains magnetised after the current is switched off depends upon the retentivity of the material. This is high for hard magnetic materials such as steel and low for soft materials such as iron and permalloy.

Magnetising characteristic
If an increasing value of current is applied to a coil surrounding a completely unmagnetised ferrous material and the induced magnetism is measured, it will be found that the relationship is not a straight-line one. The magnetic flux at first increases only slowly with increments of magnetising force; it then increases over the main part of the curve, before flattening off to the *saturation* value when the material is fully magnetised. With some substances, notably ferrous oxide (which is an important material in magnetic recording), the flux density does not return to zero when the magnetising force is removed at saturation but changes to some higher value which represents the *remnance* of the material.

Hysteresis
If an alternating current is applied to a coil so that the magnetising force acts in a cyclic manner, magnetism will be induced in alternating polarity. Because of the remnance of the material the magnetising/demagnetising characteristic does not retrace the initial magnetising curve since a force in the opposite direction (known as the *coercive* force) is required to reduce the flux density to zero. After a number of repeated reversals the curve settles down into a loop formation known as a *'hysteresis loop'* (from the Greek hysteros—coming after). This characteristic, which involves magnetising the material to saturation in alternate polarity is known as a *major hysteresis loop*. Any small repetitive reversals in the magnetising force will give rise to *minor hysteresis loops*. These appear like small versions of the hysteresis curve attached to the inside of the major loop.

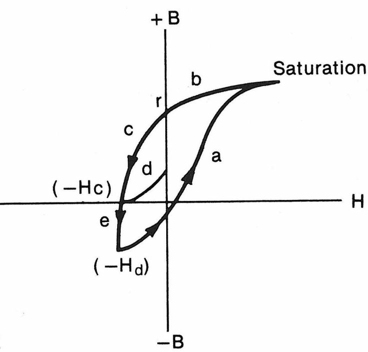
Demagnetisation
From a study of the hysteresis loop it would appear that, once saturation had taken place, zero magnetisation could never be attained with zero magnetising force. If while the alternating force is being applied it is progressively reduced in value, however, a series of diminishing hysteresis loops are produced which eventually reach zero. This process is called *demagnetisation* or, in some circumstances, *erasure*.

Variations of flux density (B) with magnetising force (H). In fine detail the curve is not smooth but is made up of a series of minute steps (called Barkhausen jumps) as the magnetic crystal elements, which form 'domains' that act like tiny bar magnets, jump into line.



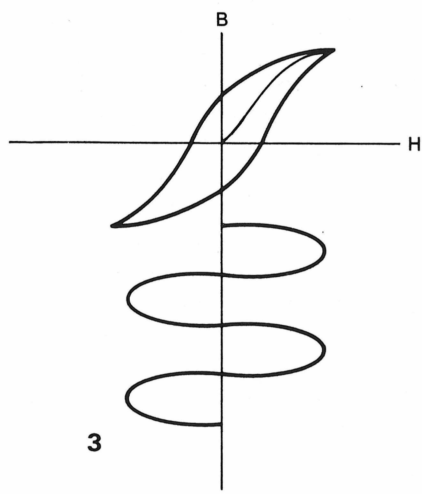
1

Curve *a* shows how the flux density in a medium increases from zero with increasing magnetic force. Curve *b* shows the effect of progressively reducing the force to zero. The flux density will diminish not to zero but to a value (*r*) which is the *remanence.*
If the sense of the magnetising force is reversed (*c*), a point will be reached where the force exactly balances the residual magnetism of the material. The value of this force ($+H_C$) needed to achieve this state from saturation is the *coercivity* of the medium.
The material is not completely demagnetised at this point and if the force were removed the flux density would rise as in curve *d.* A further increase in $+H$ is required (to $+H_d$ curve *e*) to completely demagnetise the material.



2

The initial magnetisation and subsequent variations of flux density caused by sinusoidally varying the magnetising force following many reversals.



3

Magnetic fields can be produced by passing current through an electrical conductor.

Electromagnetism

Electromagnetic fields

When an electrical current flows in a conductor the motion of the electrons sets up a magnetic field around it. This takes the form of concentric lines of force within and outside the conductor.

The direction of these lines in relation to the current is always such that they can be considered as rotating clockwise when the current is flowing away from the observer.

Magnetic coils

If a conductor is wound into a coil the lines of force will amalgamate to produce a continuous field similar to that of a bar magnet (while the current remains on) with a north pole at the end where the lines of force emerge and a south pole where they re-enter the coil. This forms what is called an electromagnet or solenoid.

Magnetising force

The strength and disposition of the magnetic field in a coil depends on the number of turns, the current flowing in them and the size and shape of the windings.

$$\text{Field strength, } H = \frac{I\,n}{l}$$

where n=the number of turns, l=length of coil and I=current. If the coil is formed into a continuous loop or toroid there will be no poles, most of the flux will be contained within the windings and the density will be practically uniform throughout the cross section.

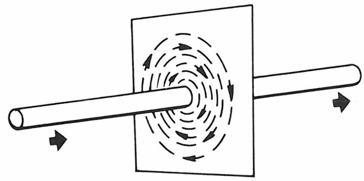
Permeability

The efficiency of a coil in terms of the magnetic effect, ie number of lines of force produced by a given magnetising force, can be considerably increased by winding it on a core of ferrous material. This efficiency (permeability) can be expressed as the ratio of the magnetic induction or flux density (B) to the field strength (H).

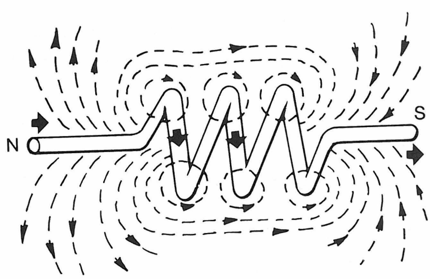
$$\text{Permeability, } \mu = \frac{B}{H}$$

In the gauss system the permeability of free space (vacuum) is taken as unity. This is known as absolute permeability, μ_0. The ratio of the permeability of other materials to absolute permeability is called the relative permeability, μ_r, and varies considerably between materials. It is very small, and substantially constant for air and for such *paramagnetic* materials as copper, aluminium etc. *Ferromagnetic* materials such as iron, steel, cobalt, etc have very much higher and more variable values of μ (orders of magnitude of 200–200 000) depending upon their composition, purity and temperature.

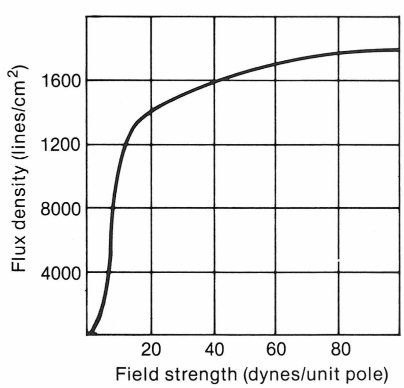
When current flows in a conductor, a magnetic field is set up around it. The lines of force flow in a clockwise direction as seen from the source from which the current flows.



If a conductor is wound into a coil, the lines of force will amalgamate to form a continuous field similar to a bar magnet.



The efficiency of a coil is increased by winding it on a core of ferrous material. The extent to which the core material improves the magnetic efficiency depends upon its *permeability,* This is the ratio of flux density to field strength. The graph shows a typical permeability curve for mild steel.



Recording Tape

The basic material for magnetic recording is tape. Magnetic disc can also be used but this tends to be reserved for specialist applications such as sound signal delay, for reverberation or PA (audience reinforcement) systems, computer memories and television action replay systems.

Tape sizes

Recording tape is manufactured in wide rolls and slit into various widths to suit different purposes. These include:

0.15 in (3.81 mm) for compact cassettes
0.25 in (6.2 mm) the standard for most reel-to-reel machines
0.5 in (1.27 mm) for multi-track audio and television recording
1.0 in (2.54 cm) for multi-track audio and television recording
2.0 in (5.08 cm) for multi-track audio and television recording

There are also 35 mm and 16 mm recording tapes with sprocket holes to match film for the synchronous recording of film sound, and the magnetic stripe which is a thin coating of magnetic material applied to one edge of a cine film on which sound can be recorded.

Base material

Recording tape is composed of two layers: the base and the coating of oxide on which the recording is made.

Various materials are used for the base. Early examples were made of cellulose acetate which was inclined to absorb moisture and dry out with storage, becoming brittle.

This has been superseded by PVC and later polyester. A tensilised polyester, which has improved tensile strength, is particularly suitable for the thin double and triple play tapes. The requirements for base materials are freedom from:

Stretch Stretching of the tape can give rise to variations of pitch in the recording.

Cupping A tendency to bend across its width, which can result in poor head contact unless pressure-pads are used.

Bias One side of the tape being longer than the other causes poor spooling.

Curl A tendency for the edges to ruffle, which can cause pitch variations.

Spooling systems

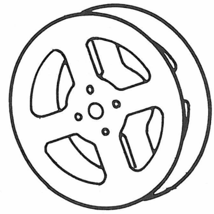
Reel-to-reel The tape is stored on a spool and is played by winding it from the storage spool to a take-up spool.

Cassette The tape runs between two spools, to which it is permanently attached, fitted into a plastic box.

Cartridge The tape is in the form of a continuous loop wound on a single spool with rollers in a single box.

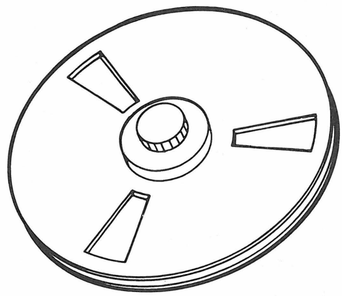
Open reel tape spools come in a variety of sizes, the most popular being 3 in (8 cm), 5 in (13 cm) and 7 in (18 cm) for domestic use. These are usually plastic 'cine' spools (so called because they can also accommodate 8 mm cine film).

1



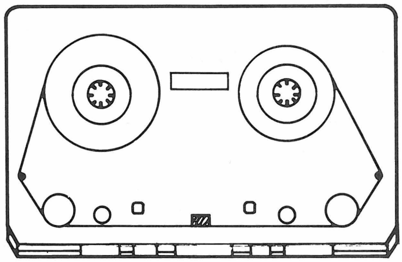
Professional recording is done on large spools. In the cases where $\frac{1}{4}$ in tape is used the spools are usually $10\frac{1}{2}$ in (27 cm) NAB spools, usually made of aluminium and sometimes with single face-plates for horizontal mounting.

2



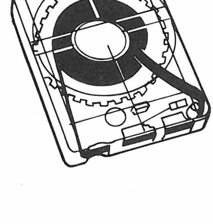
Compact cassettes use tape only 0.15 in (3.81 mm) wide permanently attached to a supply reel on the left and a take-up reel on the right.

3



Continuous loop cartirdges generally use lubricated tape.

4



The recording surface is a fine coating of ferric or chrome dioxide compound attached to one side of the base material.

Types of Tape

The base material described in the previous page is merely the vehicle to carry the recording surface which is a fine, even coating of magnetic material attached to one side of it.

Coating

The recording surface of the tape is a compound of ferric oxide, pure iron or chromium dioxide particles suspended in a 'binder' material (eg Vynylite) which adheres evenly to the base.

The oxide is produced as minute needle-like particles, each of about 0.4 × 0.04 micron. These are very thoroughly dispersed throughout the binder (in the ratio of about 70% oxide to 30% binder) and are usually magnetically aligned before the mixture hardens so that the majority have their major axis in line with the length of the tape. (In the case of 2 in tape used for quadraplex videotape recording, the particles are aligned at right angles to suit the transverse video scan; the tape is, therefore, less efficient for the lengthwise sound recording.)

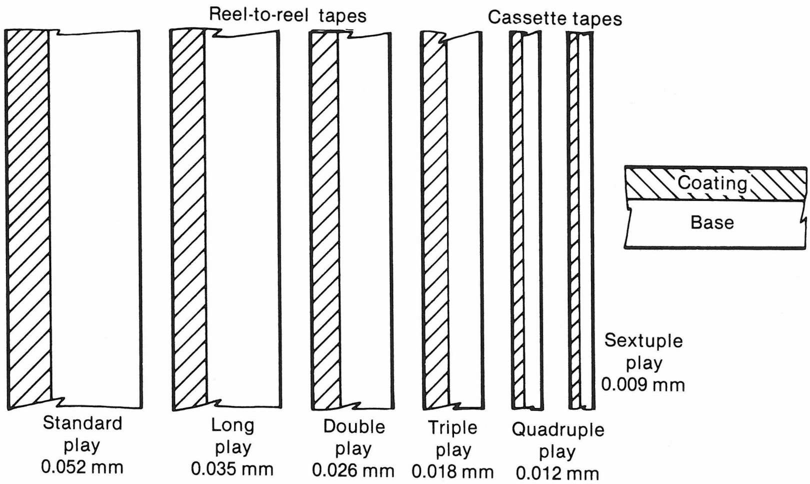
Tape surfaces

Tapes are made with shiny or matt surfaces. The matt surface usually produces better spooling characteristics than the shiny but is inclined to promote more rapid head wear. Specially lubricated tapes (eg impregnated with silicone) are available mainly for use with continuous-loop cartridges and magnetic delay systems.

Tape coating materials

A variety of types of tape are available in which the granular structure of the magnetic surface is varied to give a choice, or in some cases a combination, of three basic qualities:

1. Standard tape, which is reasonably sensitive and requires a fairly low bias current but has a rather high noise level.
2. 'Low noise' tape, which has a much better programme: noise ratio particularly as regards hiss and modulation noise. This is probably achieved by making the magnetic particles small and uniform. Unfortunately the small particles are rather easily magnetised by strongly magnetised particles on adjacent turns of the spooled tape. This gives rise to *print through*, which sounds rather like an echo effect, loud sounds from adjacent layers 'breaking through' quiet passages.
3. 'High-out' tape has a high magnetic coercivity, having a fine crystalline structure of uniform and well orientated particles. As a result it will accept a higher recording level and requires a higher level of bias. In the average domestic machine, the narrow gap of the single record/reproduce head makes it difficult to take advantage of the improved frequency response.



Thickness of recording tape. The thickness of the recording tape is the sum of the thicknesses of the base material and the coating. Different thicknesses of base materials and corresponding thicknesses of coatings are used to give different lengths of playing times. In general the thinner tapes require more careful handling and are more prone to tape-stretch (with consequent pitch and playing-time variation) than the standard play tape. They also tend to accept a lesser range of modulation without distortion but the increased playing time can be a convenience.

TYPICAL TAPE LENGTHS AND SPOOL SIZES

Spool diameter		Standard tape		Long play tape		Double play tape		Triple play tape	
cm	in	m	ft	m	ft	m	ft	m	ft
7.6	3			65	210	90	300	135	450
10.2	4			135	450	180	600	270	900
12.7	5	180	600	270	900	360	1200	540	1800
17.5	7	360	1200	540	1800	730	2400	1080	3600
25	10	730	2400	1000	3280				

PLAYING TIMES PER TRACK IN MINUTES FOR VARIOUS LENGTHS OF TAPE AND SPEED

Tape length		19 cm/s	9.5 cm/s	4.75 cm/s	2.4 cm/s
m	ft	$7\frac{1}{2}$ ips	$3\frac{3}{4}$ ips	$1\frac{7}{8}$ ips	15/16 ips
65	210	5.5	11	22	45
135	450	11	22	45	90
270	900	22	45	90	180
360	1200	30	60	120	240
540	1800	45	90	180	360
730	2400	60	120	240	480

There are various types of tape coatings, each having different characteristics.

Recording Tape Coatings

The search for improved frequency response and better signal-to-noise ratio has resulted in the production of various types of tape coatings, each with different characteristics.

Ferric oxide (Fe_2O_3) coating

The iron oxide particles (Fe_2O_3) are pulverised into needle form (about 0.5 micron long) and added, in suspension, to an adhesive compound.

Ferric oxide tape tends to have a limited high frequency response and rather poor signal-to-noise ratio.

Chrome dioxide (CrO_2) coating

Chrome dioxide particles have a finer needle shape than ferric oxide allowing for better homogenity. Chrome dioxide is distinguished by its dark colour (almost black). It provides an excellent high frequency response with the ability to record very short wavelengths (approximately 10 dB better than ferric oxide at 10 kHz). This makes it particularly suitable for slow-speed cassettes.

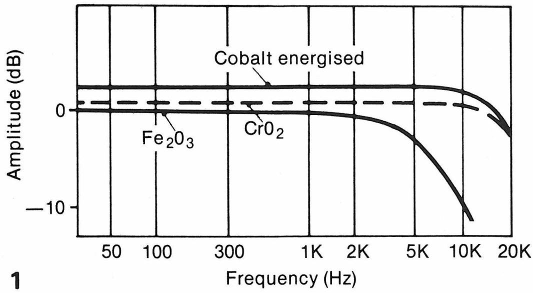
Chrome dioxide has excellent power-handling capacity in the high frequencies but a fairly high inherent hiss level. To get the best out of it, therefore, the input high frequency and bias level should be high and the replay characteristic almost a pure 6 dB/octave (see p. 108). Unfortunately chrome dioxide tape tends to produce a high degree of head wear and has a slightly inferior low frequency response compared with Fe_2O_3.

Ferrichrome ($CrO_2 . Fe_2O_3$) coating

Ferrichrome is intended to combine the virtues of the enhanced high frequency response of chrome dioxide with the low and medium frequency response of ferric oxide. A thin layer (0.04 mils) of chrome dioxide is superimposed over a ferric oxide base, approx 0.21 mils thick. Unfortunately, as the chrome layer is on the outside there is still the problem of excessive head wear due to abrasion.

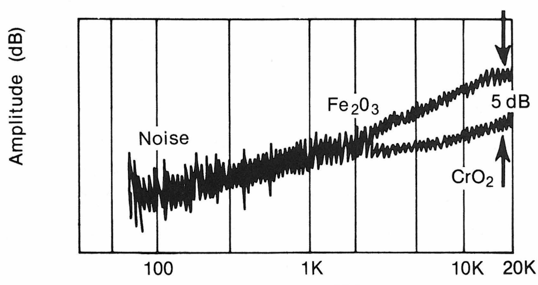
Cobalt coatings

Substantial improvements have been made in frequency response, signal-to-noise ratio and dynamic range without the need for critical bias settings, by the addition of cobalt to ferric oxide. This can be achieved by introducing an ion of cobalt into a particular formation of ferric oxide in a process similar to the 'doping' method used in the manufacture of semi-conductor materials. The surface retains the relatively smooth, soft texture of ferric oxide so that head wear is not excessive.



1

Comparison of frequency response



Comparison of noise

2

1, The graph shows a comparison of frequency response characteristics between ferric oxide, chrome dioxide and cobalt energised; 2, Comparison of noise spectrum between ferric oxide and chrome dioxide tapes.

Previously recorded material is erased from the tape by applying a powerful high-frequency alternating magnetic field.

Tape Erasure

Before a recording is made, it is usual, unless a superimposition is required, to subject the tape to a full erasure to ensure that no unwanted modulation remains on it.

For this reason a special erase head is fitted on most machines and is the first head that the tape encounters.

The erase head

The erase head consists of a coil of wire wound on a ring of magnetic material which has a small gap over which the tape passes. When erasing is required (ie normally when the machine is in the record mode), an alternating current of ultrasonic frequency (usually in the region of about 50 kHz to 100 kHz), supplied by an oscillator, is passed through the coil.

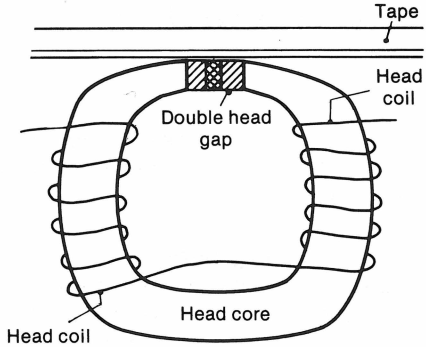
Quite a high power is involved as it is necessary to bring the tape to magnetic saturation, so the core is usually made of some very low coercivity material such as silicon steel which does not saturate readily. The core is built up in thin laminations, each insulated from the others or made of ferrite, a nonconducting magnetic material, to reduce the formation of eddy currents in the metal and consequent losses due to heat. The head gap is filled with a nonmagnetic substance such as hard copper or nickel. Eddy currents are encouraged to occur in this area as they tend to assist in the erasing process. To combat the high coercivity of modern recording tapes, many erase heads are produced with double gaps consisting of a small magnetic element sandwiched between two nonmagnetic spacers so that the tape is effectively wiped twice.

The erasing process

As the tape passes over the gap or gaps in contact with the poles, the magnetic flux circulating in the core is encouraged to divert through the tape due to its relatively low permeability. The alternating magnetic force is sufficiently strong to drive the tape around its major hysteresis loop (see p. 78) from saturation in one direction to the other.

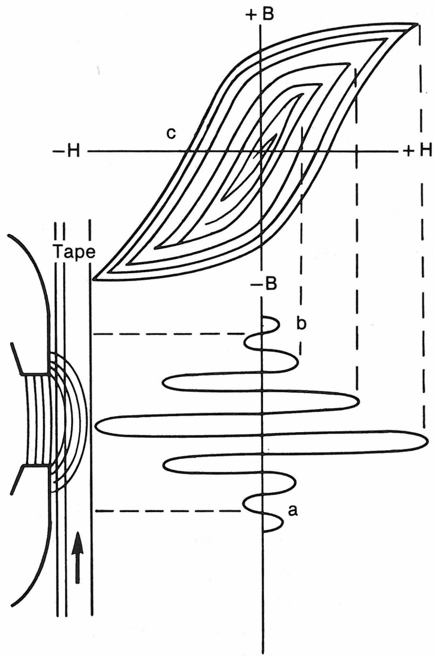
Erase heads have quite wide gaps relative to the wavelength of the bias (typically 0.01–0.05 cm) so it is possible for an element of tape to be subjected to a number of alternations of magnetism as it traverses the gap. As each element of the tape passes the head it is subjected to an alternating magnetising force which builds up to saturation and then diminishes to zero. The tape magnetisation therefore follows a series of increasing and diminishing hysteresis loops. The magnetic properties of the molecules are continually reorientated in opposite directions until, at the end of the gap, the flux is no longer strong enough to affect them. They are, therefore, left in random formation and the tape is demagnetised.

The erase head. The double head gap consists of a central magnetic section sandwiched between two nonmagnetic spacers. This has the effect of virtually erasing the tape twice thus ensuring complete erasure. An ultrasonic alternating current (supplied by an oscillator) is passed through the head coils. The current is powerful enough to take the tape to magnetic saturation in each direction. The head core is composed of finely laminated soft iron or ferrite to reduce losses due to eddy currents.



The erasing process. The tape passes through an AC magnetic field that takes it to saturation in alternate polarities. As each particle of tape leaves the gap it passes through a diminishing field so that the hysteresis loop, which has built up to saturation value, collapses upon itself and the residual magnetism is reduced to zero. The tape is thus demagnetised.

Point *a* shows the expanding flux, *b* the flux linkage diminishing and *c* the expanding and contracting hysteresis loop, reaching saturation at the mid point of the gap so as to absorb any residual magnetism from a previously recorded audio frequency signal.



The way in which electrical signals are transferred to magnetic variations on tape varies according to frequency.

The Recording Process

The purpose of recording is to impose on the tape a permanent magnetism that varies in polarity and intensity along the tape in direct relationship to the manner in which the signal being recorded varies with time.

The recording head

The recording head is a ring of ferrous material composed of ferrite, a compound with a high resistivity, or of a material with a high permeability such as *permalloy* (an alloyed iron containing silicon). The material is in granular form or built up in thin laminations, each individually insulated to reduce losses due to eddy currents.

The ring is made in two halves joined together with shims of non-magnetic material which fill the gaps at the front and back. Some high quality recording heads have quite substantial back gaps which, although they reduce the overall permeability and therefore require more driving power, make it more consistent throughout the frequency and volume range.

The poles are chamfered at the back of the gap over which the tape passes to increase the reluctance of the gap and help to push the magnetic flux out into the tape.

The head coil is wound in two sections (one on each leg) which are connected in series.

Gap size

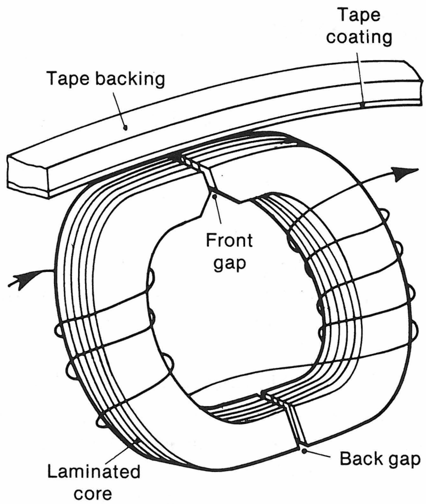
Various factors, notably tape speed, affect the choice of optimum head gap size for recording. It needs to be sufficiently large to produce a strong enough external field and deep enough penetration of the tape. On the other hand, if it is large enough to be comparable with the wavelength of the signal, loss of the high frequency response will result. This is because the magnetising force, diminishing towards the end of the gap, will cause the AC bias (p. 94) to act in the manner of the erase signal and cause partial demagnetisation.

A typical recording head gap would be of the order of 0.02 mm. Where only one head is used for both recording and reproducing, as in many domestic machines, the gap is made to suit the more stringent requirements of replay. This requires a much narrower gap of the order of 0.002 mm.

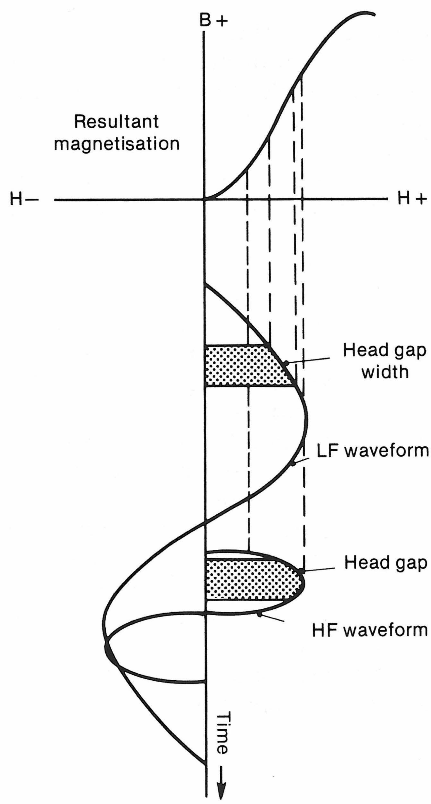
The recording process

The manner in which the recording is established on the tape is very complicated and varies with frequency. Different segments of the transfer characteristic become involved as the frequency and thus the relationship between wavelength and head gap size varies.

The recording head is made of thin laminations of ferrite or iron alloy, each insulated from the other to reduce eddy current losses. The back of the front gap is chamfered to increase the reluctance. It also helps to push the magnetic flux out so that it penetrates the tape more deeply.



The magnetisation transfer characteristic.
At low frequencies the gap represents only a small segment of the waveform so that the magnetising force is virtually unidirectional and the output tends to relate to the initial transfer characteristic. As this is non-linear, harmonic distortion results. At higher frequencies, where the wavelength is less than about eight times the gap size, reversals of direction of magnetisation will occur while an element of tape is passing the gap. This results in the formation of incomplete hysteresis loops with consequent distortion.



DC Bias

Having considered the properties of the tape and the characteristics of the recording head, we should now consider the way in which the programme signal becomes translated into a magnetic recording on the tape.

The need for bias

A study of the curve on page 79 shows that the transfer of alternating electrical energy into magnetic induction is by no means a linear relationship.

The initial magnetisation characteristic is in the form of an S in either direction from zero. If, therefore, the signal was merely applied to the recording head, severe distortion of the recorded waveform would result.

The nature and extent of the distortion would vary considerably with frequency and amplitude of the signal in relation to the tape speed and head gap size as explained on the previous page.

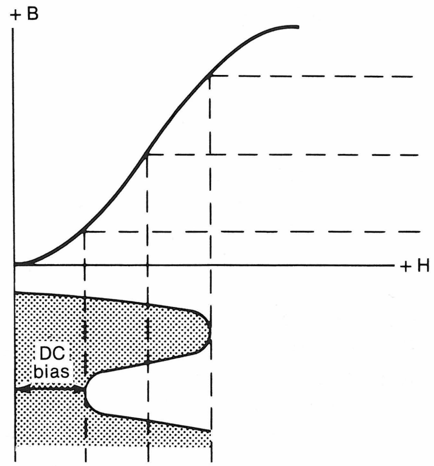
Use of DC bias

The problem of non-linearity of the transfer characteristic is tackled in some simple recording machines by the application of a magnetising force of fixed value and polarity which can be used in either of two ways:

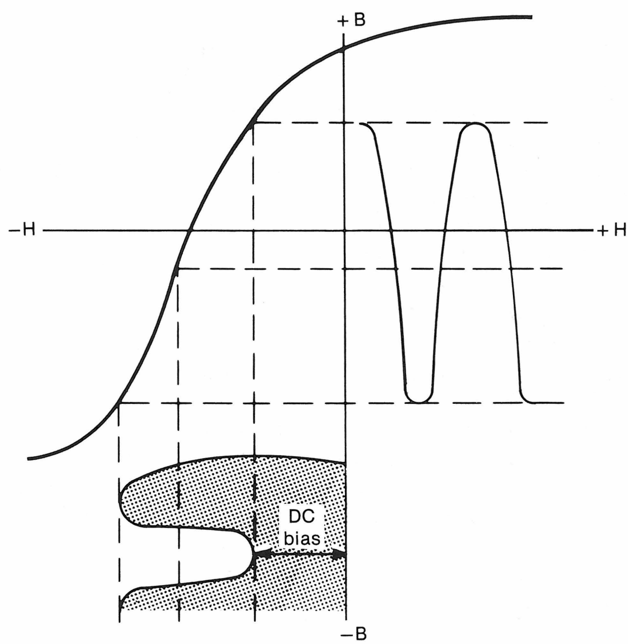
1. A direct current is superimposed on the signal to bring the operating point to the centre of the straight line portion of the transfer characteristic.
2. The tape is brought up to full saturation by a previously sited erase head (possibly even a permanent magnet) to ensure complete erasure of any previous magnetisation. The signal is then superimposed on a small demagnetising DC current so that it follows the straight portion of the hysteresis loop.

This second method has the advantage over the first one that it pre-erases the tape but both systems only work effectively for low frequencies and the noise level is very high due to the varying effect of the DC magnetisation on the random particles in the tape.

As the tape coating is composed of a variety of different sized particles orientated in random manner, it is not difficult to imagine that the DC field will produce a residual magnetism with variations matching the structure of the tape coating. When reproduced these variations will be heard in the form of a high pitched hiss. For this reason it is usual for systems employing DC bias to curtail the high frequency response.



By superimposing a small fixed DC current on the audio frequency signal, the signal can be shifted to the straight part of the transfer characteristic thus reducing distortion.



Saturation bias.
A strong bias is used to bring the tape up to saturation. This erases any previous signal recorded on the tape and sets it to its major hysteresis curve where there is room for a larger signal excursion. A larger signal can thus be recorded and the programme:noise ratio is improved.

AC Bias

We have seen (p. 92) that, due to the non-linearity of the transfer characteristic, it is necessary to apply bias to the signal when recording. The use of DC bias results in a high level of background noise. It has been found that by using an ultrasonic AC (typically about 75–100 kHz) as bias much better results can be obtained and this is used in practically all modern tape recorders.

The effect of head gap

Firstly it is important to remember that the intention of magnetic recording is to induce on to the tape a residual magnetism which varies along its length in accordance with the signal to be recorded. In other words what matters is the magnetic state of each particle of the tape as it *leaves* the influence of the head gap.

In discussing tape erasure we found that applying an AC field to the tape, which increases and then diminishes in effect as it crosses the gap, produced a series of hysteresis loops which expanded to saturation and then collapsed progressively, thereby reducing the residual magnetism to zero.

If an AC field is applied which is not strong enough to take the tape to saturation in each direction and on this AC field we superimpose a DC field, this will have the effect of shifting the general location of the hysteresis loop. As far as each individual particle of tape is concerned, as it leaves the gap the AC field of the bias will collapse not to zero but to the DC value which forms the centre of the shifted hysteresis loop. Hence the tape is left with an appropriate degree and direction of residual magnetism.

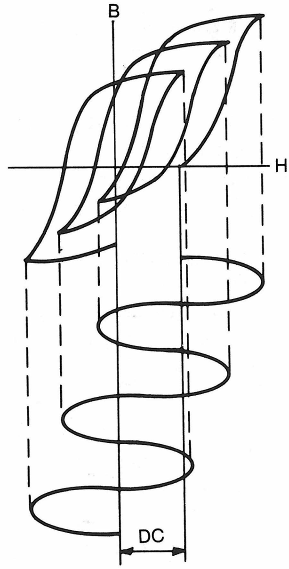
The effect of wavelength

Note that this example differs from the case of superimposing an AC programme signal on a DC bias because the wavelength of the AC bias is much shorter than the head gap size so it has time to form complete hysteresis loops as each particle traverses the gap. If, instead of the DC example, we superimpose an audio frequency signal on the AC bias, it behaves like a slowly varying DC as far as each particle of tape crossing the gap is concerned because its wavelength is long in relation to the head gap (at least 10 times longer than the bias). A residual magnetism is left that conforms with the audio waveform.

Anhysteretic curve

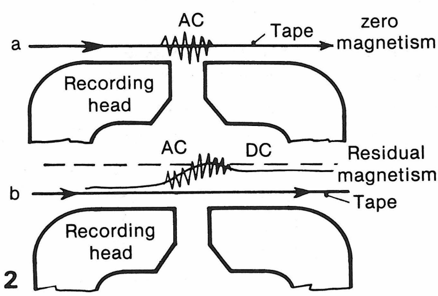
By superimposing the programme signal on an AC bias we have effectively absorbed the non-linear element of the magnetisation/demagnetisation process in the bias. The transfer characteristic now follows what is known as an anhysteretic curve.

The effect of superimposing a DC magnetism on an AC magnetising field. If the DC is suddenly applied, it takes a few alternations before the hysteresis loop settles down into its new position (displaced to the value of the DC bias).



1

A representation of the magnetising effect on the tape as it passes the recording head gap. a, If only the AC bias is present, it virtually erases itself as it leaves the gap. b, If a DC force is superimposed, the tape is left with a residual magnetism. The audio signal can be considered as a DC source in this context because, having a much lower frequency (longer wavelength) than the bias, it does not have time to execute a complete hysteresis loop while traversing the gap and leaves it with a residual magnetism which constitutes the recording.



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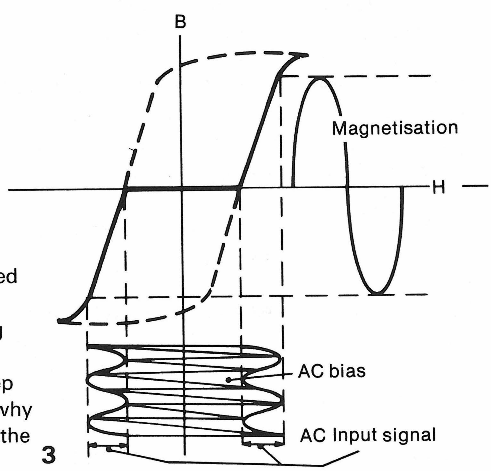
When a ferrous substance is subjected to a slowly changing DC force superimposed on an AC magnetising force, the transfer characteristic follows an hysteresis curve. The steep slope of this curve helps to explain why the use of AC bias greatly increases the reproduced output.



3

Cross-field Bias

A primary objective in tape recording is to obtain as much output from the tape relative to the noise level as possible. This requires that the magnetic field penetrate the whole depth of the coating without diminishing in intensity on the surface, where it is most effective.

Optimum conditions for recording
This optimum condition tends to occur when the recording head gap is equal to the thickness of the tape coating. A fairly high value of bias is also required. Unfortunately this combination of relatively wide recording gap and high bias tends to restrict the frequency response of the recording. From the description of AC bias (p. 94) it will be realised that the AC bias virtually wipes itself out as it leaves the head gap. If a high level of bias is used this wiping action tends to affect the audio frequency signal also at the upper end of the range where the wavelength approaches the dimensions of the head gap. This effect, which results in poor high frequency response, is a particular problem when the tape speeds are low in relation to head gap size and also tends to increase with increasing level of bias.

Use of cross-field bias
In an effort to overcome this problem and thereby to improve frequency response and programme:noise ratio at relatively low tape speeds, some machines are fitted with cross-field bias. The aim is to produce a bias field that penetrates deeply and uniformly into the tape coating and to prevent, as far as possible, the erasing effect caused by a gradual diminution of the bias as each element of the audio signal passes the head gap.

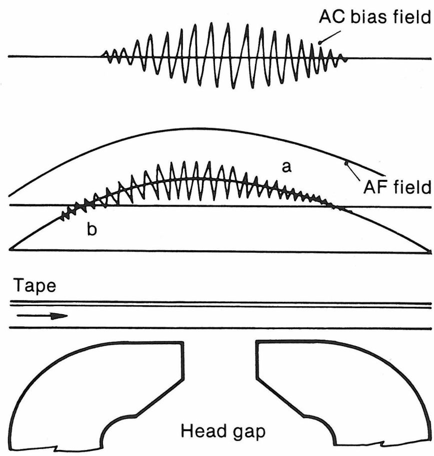
The idea is to produce a bias field that extends across the thickness of the tape and has a rapid cut-off at the point where the audio signal element leaves the gap.

Cross-field (auxiliary bias) system
One method of preventing the self-wiping action is to combine signal and AC bias in the normal way and to incorporate an auxiliary magnetic field, set up between the head and a point at the back of the tape.

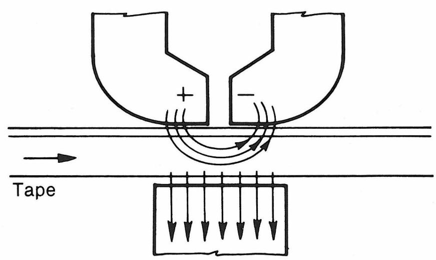
The auxiliary field is so arranged that it reinforces the bias field at the leading edge of the gap and counteracts it at the trailing edge.

Cross-field (separate bias) system
An alternative method is to feed the signal to the recording head without bias, which is applied by an entirely separate bias head operating through the back of the tape. The two head gaps are offset so that the bias is applied to the tape first and is diminishing at the point where the programme signal is applied.

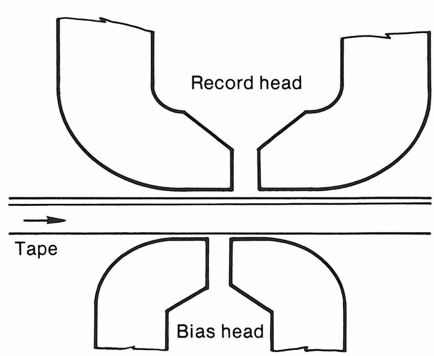
Cross-field bias is intended to produce good flux penetration of the tape and reduce the 'wiping' action of the bias as it affects the high audio frequencies. With normal bias the tape is passed over the head gap where it comes under the influence of the AC bias field. The curve *a* represents the resultant magnetic effect on a particle of the tape as it traverses the gap. The diminishing bias field at *b* is tending to 'wipe' the AF field thereby reducing the recorded level.



The use of an auxiliary field which reinforces the bias field at the leading edge of the gap and counteracts it at the trailing edge.



Using a separate head for cross-field bias. The bias head is positioned ahead of the audio head so that the bias field is diminishing by the time the programme signal is applied.



Optimum adjustment of bias is an important factor in producing high quality recordings.

Bias Adjustment

The correct adjustment of bias level is an important prerequisite of the making of a good recording. It should be checked at regular intervals and re-adjusted when different types of recording tape are used if, as is often the case, their bias requirements differ.

Effect of bias level on output

The strength of the bias current controls the recording sensitivity and therefore has a bearing upon the eventual replay output.

If the relationship between bias current and replay output are plotted for a constant level input, it will be found that the sensitivity increases rapidly with current to a peak value at which it levels off and then progressively decreases. This is due to the fact that the field laid down by the recording head and the sensitivity of the replay head is greatest at the surface of the tape and falls off with distance, ie through the depth of the tape.

Overbiasing the recording head tends to push the region of maximum signal intensity deeper into the tape so that the flux available to the replay head diminishes.

Effect of bias on distortion

To assess the effect of bias on distortion it is necessary to consider this factor in relation to the output level. Maximum undistorted output is defined for this purpose as the output produced by a pure tone in which the third harmonic content is 3%. This condition occurs when the bias is rather larger than that which produces maximum output.

Effect of bias on frequency response

Increasing bias current beyond the value corresponding to optimum output has a progressively adverse effect on the high frequency response. This can be due to the effect, mentioned earlier, of the area of maximum magnetisation being driven deeper into the tape. The high frequency response is particularly susceptible to separation between tape and head gap (see p. 000).

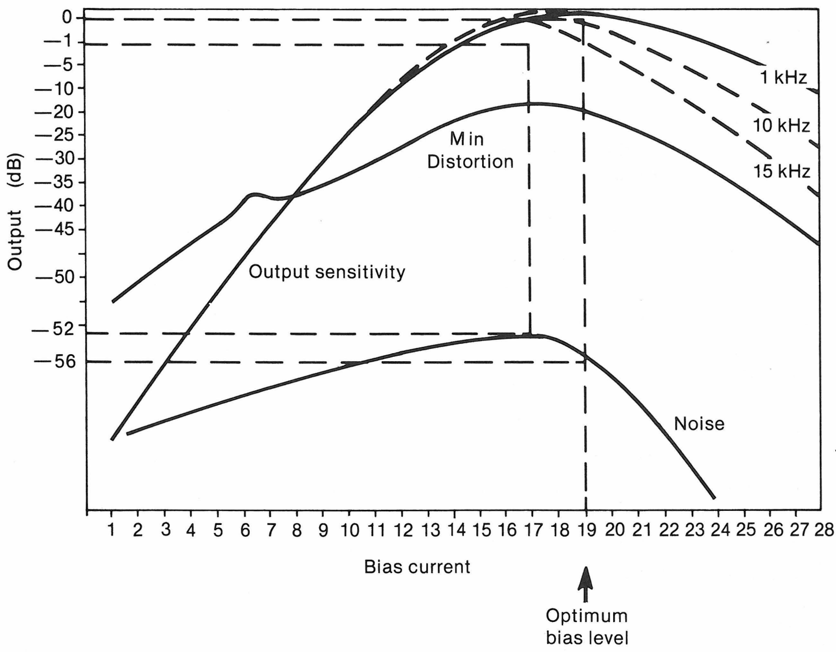
At very high frequencies, where the wavelength is comparable to the gap length, the bias can produce partial erasure as it fades toward the edge of the gap (see p. 94).

Effect of bias current on noise

Tape noise tends to increase with bias current to a point similar to that of maximum sensitivity and then decline sharply.

Choosing bias current

Taking account of all the factors, it is evident that the optimum value for bias is slightly greater than that which produces maximum output.



Curves indicating the effect of various values of bias current on output sensitivity, distortion (maximum undistorted output) and noise.
The values for bias amplitude are arbitrary units; the actual level depends on the properties of the tape, the construction of the head etc.
Superimposing the curves makes it evident that the optimum condition for bias occurs at a level just above that which provides maximum output. The method for adjusting bias, therefore, is to set it for maximum output, using standard level tone (1 kHz), and then increase it until the level drops by 1–2 dB. The higher figure gives a measure of insurance against wide fluctuations in level caused by variations in the tape coating. A better system is to use 10 kHz tone, where this is available, and adjust for a 3 dB drop from maximum output with increasing bias. This is because the steeper fall of the 10 kHz curve makes it easier to judge the correct adjustment.

The replay head must have a very narrow gap for good high frequency response.

The Replay Head

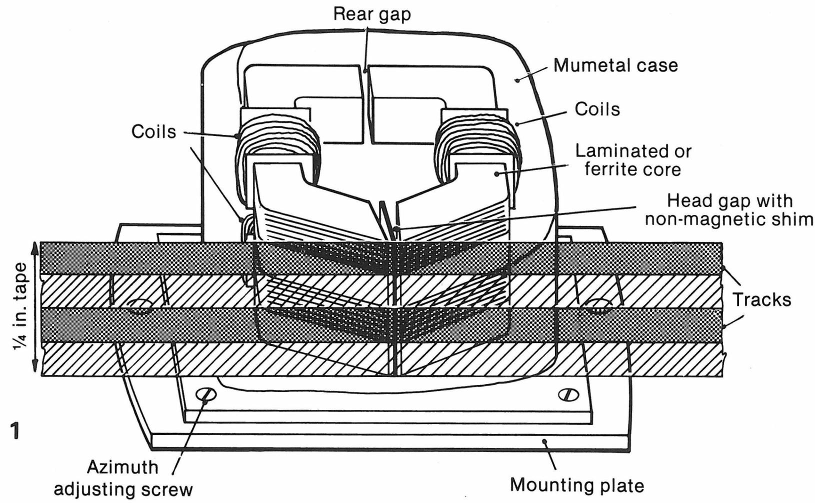
The replay head is very similar in construction to the recording head; in fact many domestic tape recorders employ the same head for both purposes. This calls for a certain compromise in design but as the requirements for replay are the more stringent this tends to be the prime consideration. The aim is to obtain a high output from the tape consistent with a sufficiently extended frequency response.

The core
Ring-type cores are used. They are composed either of high permeability material made up in thin laminations (of the order of 0.012 cm or less) each insulated from each other or of a ferrite material (a compound containing particles of iron compressed into shape) which has high resistivity. In either case the purpose is to reduce losses due to the formation of eddy currents. These are electrical currents, induced in the core by the fluctuating magnetic field, which would tend to waste energy through the production of heat. The cores are manufactured in two halves which are then clamped together resulting in a gap at the back and at the front.

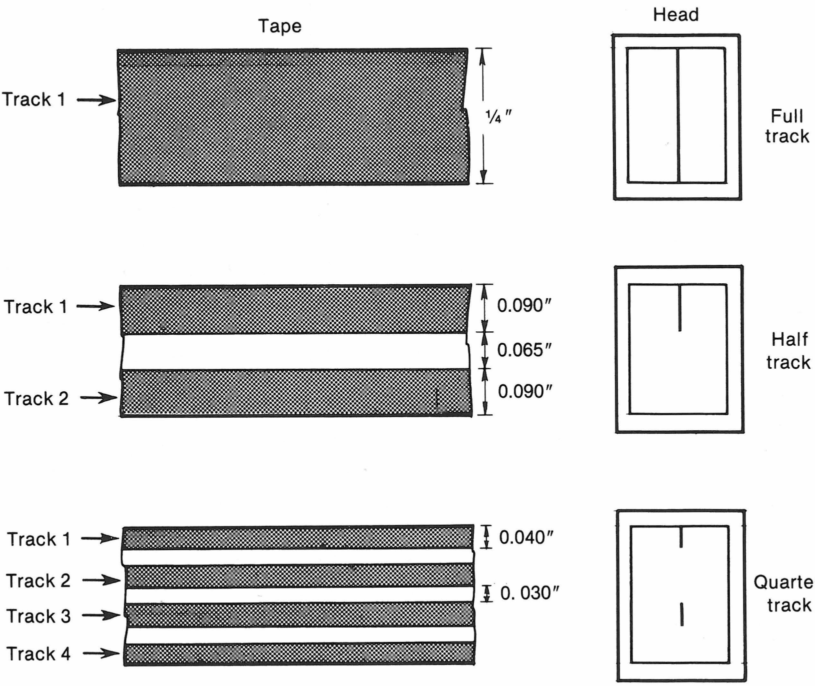
Back gap
The effect of the gap at the back of the head is to reduce the overall permeability characteristic and thereby the output. It does make it more consistent, however, and thereby improves the frequency response. The output from the head can be improved by interleaving the laminations across the back gap.

The coil
The coil is usually wound in two sections, one on each leg of the core. They are wound astatically, ie connected in series in such a manner that the current caused by induction within the core adds up, whereas that from external fields cancels out. The whole is enclosed in a mumetal shield to reduce interference from external magnetic fields to a minimum.

The front gap
The width of the front gap is related to tape speed in determining the frequency response. As explained on page 90, the high frequency response is limited by the width of the head gap. If a good high frequency response is to be obtained with a relatively low tape speed the gap must be very small. Gap sizes of 0.005 mm or less are common, reducing to 1 μm for some cassette recorders with tape speeds of 4.75 cm/sec. The lower limit of gap size is conditioned, apart from practical considerations, by the need to maintain sufficient output at the low frequencies where the proportion of the waveform, and thus the available output, across the gap is reduced.



1, Dimensions and positions of tracks on $\frac{1}{4}$in tape.



2

2, A cut away view of a quarter-track head. (a) Mumetal case. (b) Laminated, or Ferrite, core. (c) Coil. (d) Tape. (e) Position of tracks. (f) Head gap (filled with paramagnetic shim.) (g) Mounting plate. (h) Azimuth adjusting screw. (See page 102.)

Tape Head Mechanical Adjustments

To obtain the optimum performance from a tape recorder it is necessary to align the recording and replay heads very accurately. There are four important adjustments: height, zenith, wrap and azimuth.

Height
Height can be checked visually by equalising the distance between the edges of the tape and the top and bottom shield plates. For aligning stereo heads, a test tape is available with a signal recorded only in the guard band. The height is adjusted for minimum signal in both tracks.

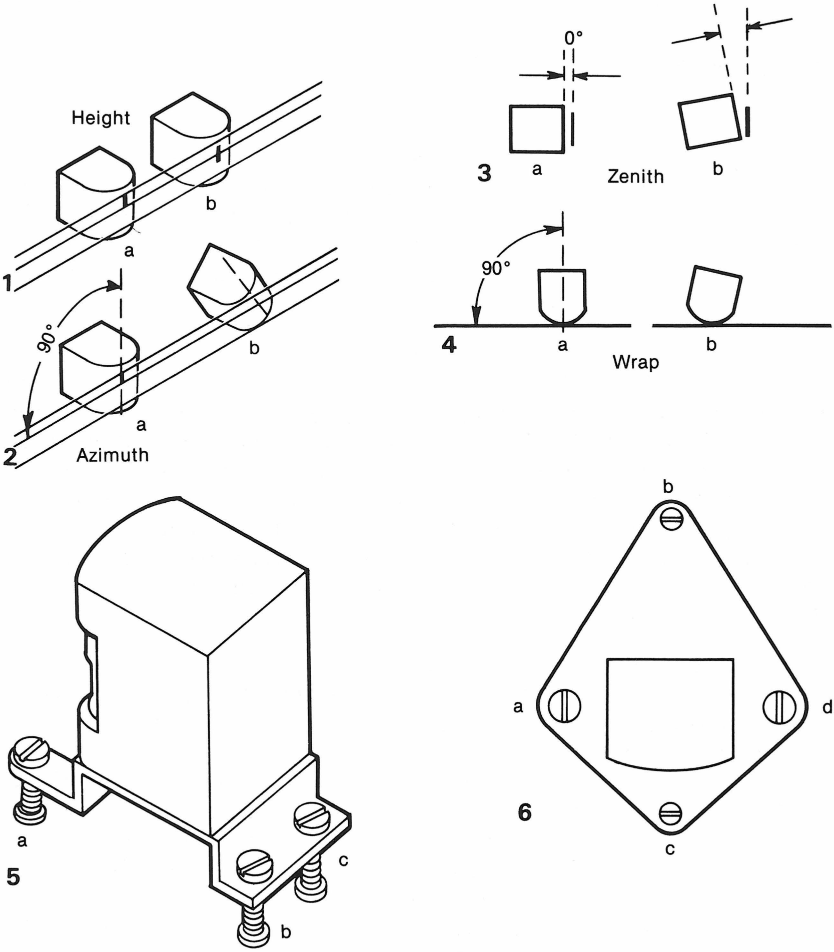
Zenith (vertical alignment)
Zenith is the extent to which the head is exactly perpendicular to the deck and parallel with the tape. Incorrect zenith results in uneven wear on the head and can cause the tape to slide up or down the head affecting track alignment.

Zenith can be checked by smearing the head with ink and observing the wear pattern after playing a short length of tape. If the deck is unobstructed, the head can be aligned with a set square.

Wrap (horizontal contact angle)
Wrap is the measure of the evenness with which the tape wraps around the head. To be correct the head gap must be at exactly 90° to the longitudinal dimension of the tape. This adjustment only affects high-quality machines that do not employ pressure pads. It can be checked by recording or playing back 16 kHz tone and lightly touching the supply spool. If the signal increases the wrap is incorrect. The head should be adjusted to give maximum output at 16 kHz and minimum change with increase in back tension.

Azimuth
Azimuth is the adjustment to set the head gap at right angles to the direction of motion of the tape. Incorrect azimuth will not be noticeable when playing back on the machine on which the recording was made but playback of a correctly recorded tape will result in loss of high frequency. Head azimuth is set with a special section of the test tape. It is adjusted for maximum output.

An alternative is to display two tracks on a dual-beam oscilloscope and adjust for matching phase.



1, Height adjustment: *a* correct; *b* incorrect (head gap is not correctly aligned with recorded track on tape); 2, Azimuth: *a* correct (head gap at 90° to length of tape); *b* incorrect; 3, Zenith: *a* correct (vertical force of head in parallel to tape); *b* incorrect; 4, Wrap: *a* correct (lateral angle of head is in line with tape motion); *b* incorrect; 5, Typical head adjusting mechanics. The spring-loaded screw *a* adjusts height and Zenith. Screws *b* and *c* adjust height and azimuth. The screw holes may be slotted to adjust for wrap; 6, Alternative arrangement. Screw *a* is spring-loaded and serves as a pivot. Height and zenith are adjusted with screws *b* and *c* and azimuth with screw *d.* With most systems the adjustments are interrelated.

The Replay Process

In the preceding pages we have considered the process of magnetic recording, ie the conversion of a varying electrical signal into corresponding variations of remnant magnetism along the length of the tape. In the replay process we have to 'read' these variations of magnetisation of the tape as it passes the replay head and reconvert them to electrical signals.

The recorded signal
The recorded tape can be considered as a collection of tiny bar magnets laid end to end along the length of the tape. They vary in length according to the frequency of the programme signal recorded at that point on the tape, each corresponding to half a wavelength. The polarity of the field in each magnet is related to the direction of the current that originally induced it, so that a complete cycle of the original waveform would be represented by two magnets with like poles adjacent (north to north and south to south according to the phase of the waveform).

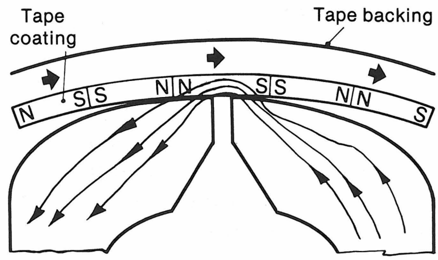
The external magnetic field
As the magnets have open poles the magnetic flux extends outside the magnets and, in fact, outside the surface of the tape, in the form of loops between the poles.

If a tape is recorded with varying frequency but the magnetising force is kept constant a series of different length magnets will be produced, each with the same total flux but with a concentration (flux density) that increases as the magnets get smaller, ie the flux per unit length of tape increases with frequency.

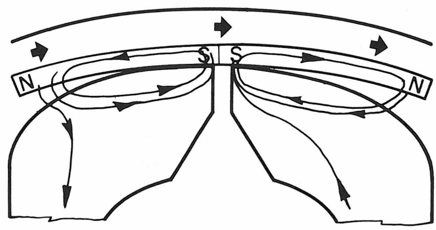
It is these external fields which link with the coils in the replay head as the tape is drawn past it and produce the output by induction. Being proportional to the rate of change of flux, the output increases as the flux per unit length of tape increases. The output therefore tends to be directly proportional to frequency, rising at a steady rate of 6 dB per octave over much of the frequency range. The output characteristic at each end of the usable frequency range is, however, modified by the physical characteristics of the replay head (see p. 100).

Tape–head contact
Because the field from the shorter magnets is more concentrated, it has a smaller spread than that produced by longer (lower) frequencies. The high frequencies are therefore particularly susceptible to separation between the tape and the replay head—hence the need for regular head cleaning to maintain good high frequency response. This fact also explains why cross-talk between adjacent tracks, that have been recorded with insufficient spacing, tends to favour the lower frequencies.

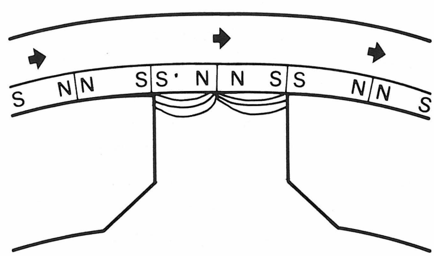
A diagramatic representation of replay head gap showing the head core and gap. The illustration shows a mid-frequency recorded on the tape at the point of a peak of a positive half cycle.



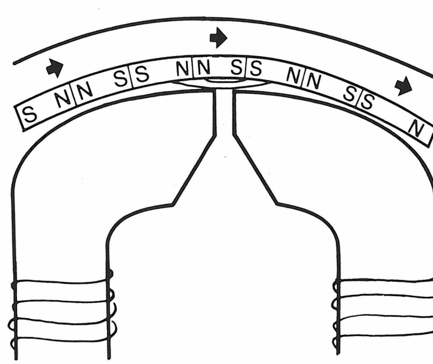
Illustrating low frequency losses due to the fact that the wavelength is longer than twice the head/tape contact area. This results in some of the flux bypassing the ring of the core.



The cancellation of output due to wavelength is equal to the gap length. (Extinction frequency.)



The effect of an impurity across the gap of a replay head. The magnetic flux from the tape is effectively diverted, thus reducing the output.



A flat overall frequency response for a magnetic record/reproduce system can only be achieved by modifying the characteristic at various points in the process.

Record/Reproduce Characteristics

As in any record/reproduce system, the object of tape recording is to produce a faithful reproduction of the original signal with nothing added, subtracted or altered. Although a straight-line frequency response is required overall, however, it cannot be maintained throughout the process because of the fundamental laws of electromagnetic induction.

Replay characteristics

The tape recorder replay head responds to the *rate of change* of the magnetic fields on the tape. If, when the recording was made, the current in the recording head was kept constant while the frequency varied, the magnetic flux induced into the tape would be constant (subject to various limitations discussed later). For most of the frequency range, however, the flux per unit length along the tape increases as the frequency increases. As the tape is drawn past the replay head, the rate of change of flux and therefore the electromagnetic induction and the output increase in proportion to frequency. Over most of the usable range the rise is directly proportional to frequency, ie 6 dB per octave.

Extinction frequency

This characteristic continues up the frequency scale until a point is reached where the wavelength of the magnetic variations on the tape (which depends upon frequency and tape speed) is small enough to become comparable to the size of the replay head gap and the output falls off dramatically. When the wavelength and the head gap are the same size, both halves of the cycle occur across the gap and complete cancellation occurs. This is called the 'extinction frequency'.

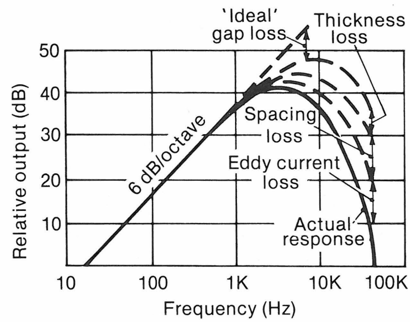
Spacing loss

In practice, the replay head output does not maintain its theoretical 6 dB per octave characteristic as far up the scale as its dimensions would predict; neither does the extinction frequency occur at exact multiples of head gap size, due to some inevitable separation between the tape and the head gap. This loss is directly proportional to the separation and to the frequency.

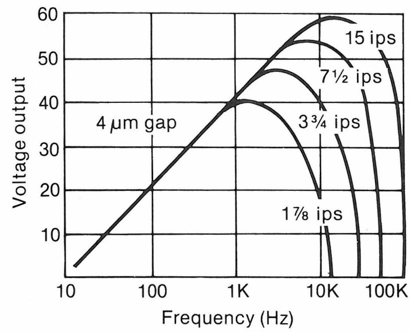
Thickness loss

The thickness of the tape can also add a proportionate spacing loss at high frequencies because, apart from that proportion of the total flux that is present on the surface, the flux in the inner areas is spaced away from the head gap by the thickness of the tape.

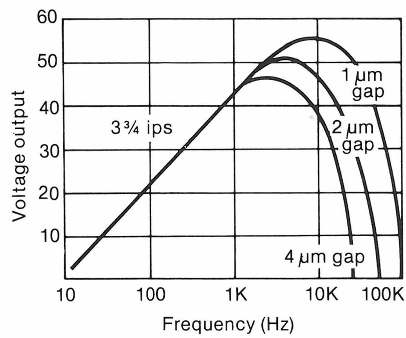
Illustrating how the basic 6 dB per octave playback curve is modified by thickness loss, spacing loss and eddy current loss.



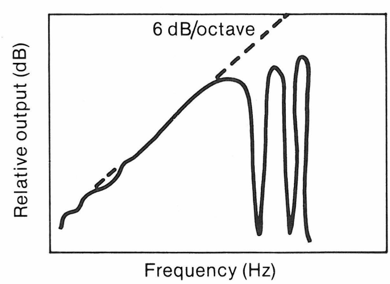
Showing how the high frequency response varies according to the tape speed.



Showing how the high frequency response varies according to head gap size, for a speed of $3\frac{3}{4}$ ips.



Recording characteristic showing general rise of 6 dB per octave, with uneven low frequency response due to magnetic flux loss. Note that extinction frequency minima occur below the frequencies where gap size is an exact multiple of wavelength. This is due to spacing loss which effectively increases gap size.



The non-linear response of the magnetic record/replay process makes it necessary to employ equalisation to both functions.

Equalisation

On the previous page we showed that the output of a tape replay head has a response which rises proportionately with frequency over most of the usable range and falls off rapidly at each end of the scale. In order to reproduce the original signal faithfully it is, therefore, necessary to apply correction by altering the response of amplifiers at various points in the process.

Method of correction

The process of correcting for an uneven frequency response is called equalisation. It could be applied in various ways to overcome the basic non-linearity of the magnetic recording system. For instance, one approach to the problem of increasing output with frequency would be to supply the recording head with a signal current inversely proportional to frequency over the major part of the scale; the recorded flux density would then be linear and the replay output flat. In fact, this is not practical as the tape would not accept the necessary dynamic range.

Replay correction

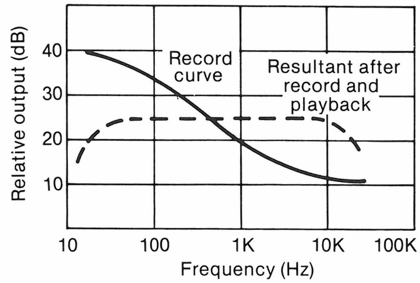
Most of the correction is, therefore, applied during replay. The equalisation is basically the inverse of the 6 dB per octave rise inherent in the magnetic induction process, with the curve flattening off at the top end of the scale to compensate for the high frequency losses (see p. 106). The amount of high frequency boost that can be applied during replay is, however, limited by the increase in the level of tape hiss that would result.

Recording characteristic

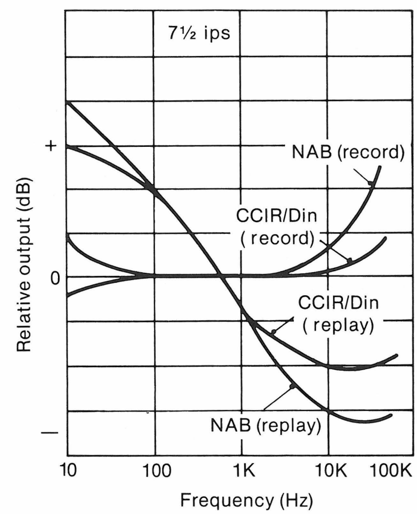
The recording characteristic is basically flat over most of the range but sufficient boost is applied to the high frequencies in recording to make good the losses. These are due to self-demagnetisation, decrease in permeability of the recording head and tape, eddy current losses, tape thickness and the erasing effect of the high frequency bias current.

In one system (NAB, described on p. 110) the low frequencies are also enhanced during recording to compensate for the low low frequency sensitivity of the replay process (when the rate of change of flux across the head gap is very small), and thereby improve the programme: noise ratio.

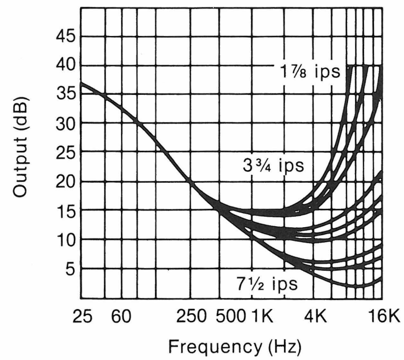
Further equalisation can also be applied to take account of different tape coatings.



Overall record/reproduce characteristic.



Recording and replay characteristic curves.



Replay curves showing differing responses for different head gap widths for speeds of $1\frac{7}{8}$ ips, $3\frac{3}{4}$ ips and $7\frac{1}{2}$ ips.

The use of standard equalisation curves enables recordings made on one machine to be reproduced faithfully by another.

Recording Standards

The process of correcting for non-linear frequency response by applying the inverse characteristic is known as equalisation.

Standard equalisation

In the case of recording systems it is important to lay down standards of equalisation so that recordings made on one machine are playable on another. The standards used are generally those developed by the National Association of Radio and Television Broadcasters NARTB (or NAB) or the DIN standards which are modifications of the International Radio Consultative Committee (CCIR) standards by the German Standards Organisation.

Time constant

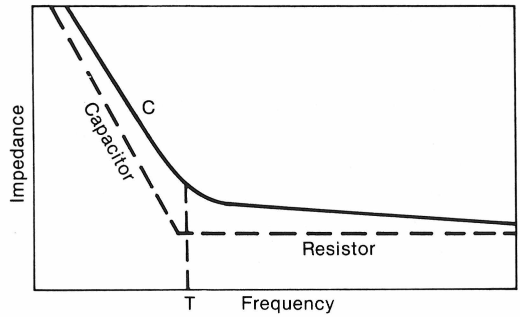
The method of stipulating the prescribed curves used in the international standards is to relate them to the variation in impedance or admittance (reciprocal of impedance) with frequency of a simple resistance and capacitance network. The actual component values are not given but their product (capacitance×resistance) is quoted in the form of a time constant, in microseconds.

The significance of this will be appreciated by considering the action of a simple capacitor/resistor series network. At low frequencies, when the capacitive reactance is relatively high, the resistance can be ignored and the admittance increases almost in proportion to frequency: at high frequencies the capacitive reactance can be ignored and the impedance becomes mainly restive and virtually independent of frequency.

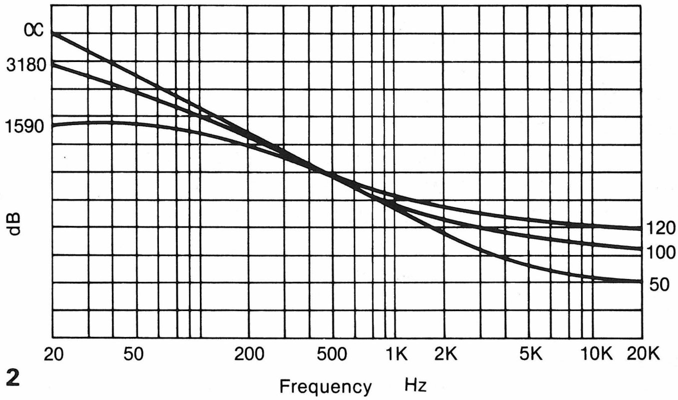
The combination thus produces a curve that represents a smooth transition between a nominal 6 dB per octave slope and linear, centred about the point where the 'idealised' curves of their individual impedances would meet. The 'turn over' frequency is the point at which the curve deviated 3 dB from the straight line.

Tape standards

Tape recorder standards are set by means of a standard recorded tape on which a full frequency run has been recorded, with such a characteristic that the curve of remanent flux on the tape versus frequency obeys a definite law. Different laws are specified for different tape speeds and there are variations between the various standards. These characteristics can be specified in terms of time constants. Where it is necessary to adjust the frequency response at the bass and treble ends of the spectrum, time constants are quoted for each turn-over frequency. Playback equalisation is designed to reproduce a linear output from these curves, in relation to the selected speed, taking account of the replay head characteristics.



1



2

STANDARD TAPE REPLAY CHARACTERISTICS AND TIME CONSTANTS

	4.75 cm/sec		9.5 cm/sec		19 cm/sec	
Standard	Bass	Treble	Bass	Treble	Bass	Treble
CCIR	—	140	3180	120	3180	100
DIN	1590	120	3180	90	3180	50
NARTB	—	—	3180	100	3180	60

1, Curve relating impedance to frequency in a circuit consisting of a resistance and capacitor in series. The resistor line represents the impedance of the resistive element which is independent of frequency. The capacitor line represents the impedance of the capacitor, which is inversely proportional to frequency according to the equation $Z=1/2\pi fC$, where Z=impedance, f=frequency and C=capacitance. This curve reduces at the rate of approximately 6 dB per octave. If the two impedances are connected in series the result will be a curve as in *c*. The 'turn-over' point or 'corner frequency' (ie the point on the curve nearest to where the two 'idealised' curves would meet if they continued in a straight line) occurs at $T=1/2\pi f$ where T is the time constant. The time constant (in seconds) is the product of the resistance (in ohms) and the capacitance (in farads) $T=C\times R$. In practice, because of the size of the units involved, the time constant is normally expressed in microseconds; 2, Time constants are a very convenient way of describing the shape of response curves. The graph shows theoretical tape replay characteristics with their bass and treble turn-over frequencies.

The tape must be made to pass the record/replay heads at a constant and accurate speed.

Tape Transport

A primary requirement for any tape recording system is a means of making the tape pass over the heads at a constant, steady speed. When replaying, the tape must pass the heads at exactly the same speed as that at which it was recorded. To reproduce tapes recorded on other machines, the speed must conform to one of the internationally agreed standards.

Standard speeds

The speeds in common usage are: 15 in/sec (38.1 cm/sec), $7\frac{1}{2}$ in/sec (19.1 cm/sec), $3\frac{3}{4}$ in/sec (9.5 cm/sec), $1\frac{7}{8}$ in/sec (4.8 cm/sec) and $\frac{15}{16}$ in/sec (2.4 cm/sec).
The high speed, 15 in/sec and in some cases 30 in/sec, is used for professional purposes where high quality and ease of editing are more important than saving tape. $7\frac{1}{2}$ and $3\frac{3}{4}$ in/sec are the speeds normally available as alternatives on domestic reel-to-reel machines. $1\frac{7}{8}$ in/sec is the standard speed for most compact cassette systems. $\frac{15}{16}$ in/sec is used for situations where length of play is more important than quality, eg 'talking books', transcription for content checking and dictation.

Speed variation

The annoyance caused by inaccurate recording/replay speed varies between individuals. Some people who are gifted with 'perfect pitch' can detect quite small inaccuracies which would be unnoticed by the majority. What is obvious to all is any form of pitch variation particularly if it is cyclic, or regularly recurring, in nature. A pitch change of a quarter tone, ie of the order of 3%, is obvious to most people if it occurs in the middle frequency range.

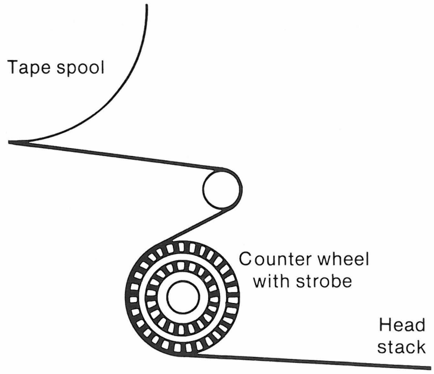
Wow and flutter

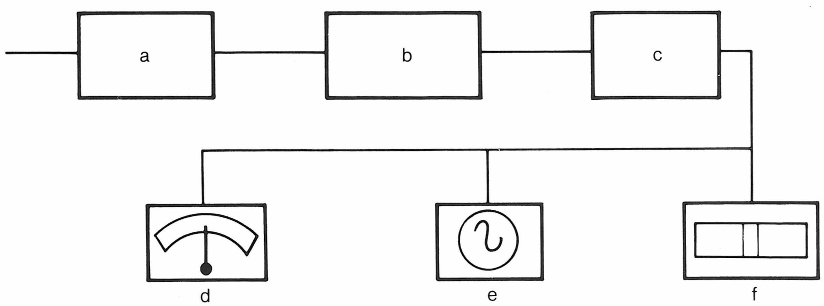
Speed variations with a repetition rate slower than about 10 per second are termed 'wow'. They produce a 'wavey' effect that is most noticeable on long sustained notes. Rapid variations (above about 10 per second) are called 'flutter'. They create a roughness of sound rather akin to the beat effect produced when two flutes are played in thirds.

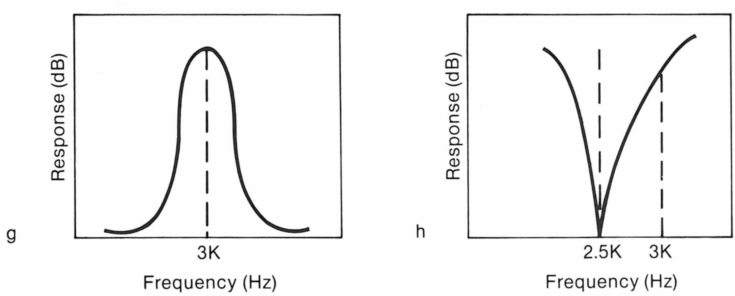
Wow and flutter are usually expressed as a percentage, indicating the variation with respect to the long term speed.

If the recording and replay are done on the same machine the effect can be doubled (or cancelled) so that in order to keep the wow and flutter below detectable limits (about 2%) a wow and flutter figure of less than 0.1% should be specified. A figure of 0.5% is considered to be the upper limit of tolerance for long term speed variation.

Some tape machines incorporate a strobe in the tape path, usually on the tape counter wheel, otherwise a portable strobe can be used (see p. 135).
When viewed by 50 Hz AC lighting, the appropriate strobe band should appear stationary. Any rhythmic variation suggests wow.







A means of measuring wow and flutter. A 3 kHz tone is recorded and played on the machine through a narrow band amplifier (*a*) with a response as in graph (*g*), a limiter (*b*), a discriminator (*c*) a wow meter (*d*), a flutter meter (inertialess) (*f*) and/or an oscilliscope (*e*).
If the discriminator is tuned to a frequency close to the 3 kHz working frequency it will produce a frequency-modulated output that varies proportionately with any frequency fluctuations, ie wow or flutter. The purpose of the limiter is to prevent variations in recorder output, due to tape variations etc, from giving false readings.

The primary requirement of a tape recorder is a constant speed capstan drive motor.

Tape Recorder Motors

Motor speed

One of the most important aspects of tape recording is steady and constant speed.

The tape transport mechanism must be capable of pulling the tape passed the heads at exactly the correct speed, so that tapes recorded on one machine can be played back on another without changing the pitch (audio frequency) of the recorded material. The movement of the tape must also be steady and constant so that the output does not suffer from 'wow' (slow cyclic variations in speed) (p. 112).

Induction motors

The prime factor in achieving constant speed is the capstan drive motor. Domestic grade tape recorders generally use AC induction motors in which the speed of rotation is related to the frequency of the mains supply.

In its simplest form the induction motor consists of a stator formed by winding coils around a series of laminated cores. Inside is the rotor which is cylindrical in shape and wound with a number of conductors in the form of closed loops. When an AC voltage is applied to the stator coils it produces a magnetic flux which induces a current in the rotor coils by induction (there being no direct connection). This produces a magnetic flux in the rotor which interacts with the stator flux and causes it to rotate.

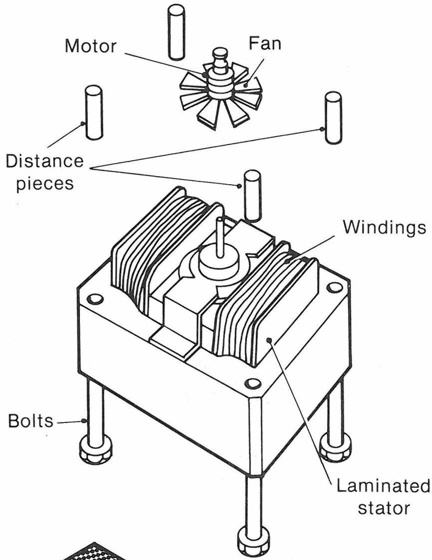
Once the motor is run up, it locks to a speed determined by the relationship between the supply frequency and the number of poles on the stator. The larger the number of poles the slower it will turn but the greater the torque and speed stability. The better machines, therefore, tend to have multi-pole motors driving the capstan spindle direct, without reduction drives; speed change is affected by switching the number of poles.

DC motors

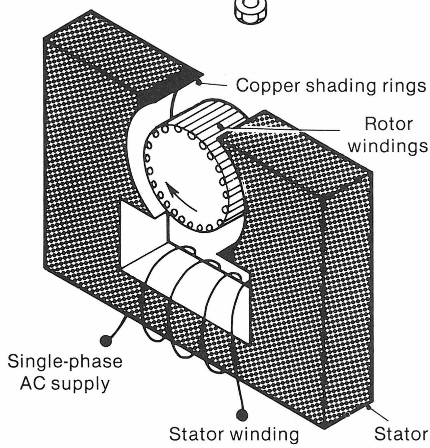
The cheaper portable recorders use simple DC motors with rotating commutators and carbon brushes fed from a stabilised battery supply. Professional grade portable machines and high grade static machines use brushless (transistor switched) DC motors. Their speed is controlled by a servo system which compares the frequency of a waveform derived from a stroboscope tachometer on the motor with that of a stable oscillator, usually controlled by a quartz crystal.

External reference

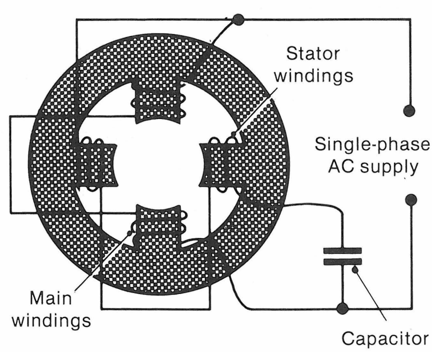
Some tape recorders provide the facility to lock the capstan speed to the frequency of an external oscillator for synchronising purposes or to achieve wide variation of speed.



Exploded view of a simple induction motor as used in some inexpensive tape recorders.



Shaded-pole motor.
Single phase AC motors will not start to run unaided. One method is to incorporate copper rings into the stator laminations. This creates some 'slippage' in the magnetic field so that the motor does not 'lock up' solidly. The magnetic flux changes at mains frequency and the rotor never quite catches up. The speed therefore varies with load. These motors are thus more suited to spooling than capstan drive applications.



Capacitor start.
An additional set of stator windings is arranged at 90° to the main windings. These are fed through a capacitor which changes the phase through 90°. The current through the two windings, differing in phase by 90°, sets up a rotating field which the rotor follows.

Constant tape speed is maintained by pinching the tape between a flywheel-driven capstan and pinch-roller.

Reel-to-reel Tape Transport Systems

The tape must be pulled past the heads at a constant speed regardless of the amount of tape on the spools. This is achieved by pinching the tape between a constantly rotating capstan and a rubber or neoprene pinch-wheel.

The capstan has to be carefully engineered to be perfectly circular and concentric, with a finely polished surface and connected to a heavy flywheel. It should be held in a long bearing, to withstand the side thrust of the pinch-wheel which is spring-loaded against it. The flywheel helps to provide the surge of energy required to pull the tape up to correct speed almost instantaneously when the load is suddenly applied. It also helps to even out the rather pulsating power supplied by the usual single-pole induction motor. Additional features to even out the fluctuations of power are usually built into the drive between motor and flywheel. This may be either a spring or rubber belt drive or a rubber-tyred jockey wheel. The motor itself is usually mounted in rubber buffers.

The facility of speed selection is achieved in some models by varying the motor speed by electrical switching but the most common method, used in the majority of domestic tape recorders, is to change the drive ratio between motor and flywheel, eg by moving the belt on to different sizes of pulley wheel.

Tape head contact

During recording or replay the tape must be made to lie as close as possible to the gap but without too much pressure if the tape and head wear are not to be excessive. Any movement of the tape away from the heads will cause a loss of high frequency response.

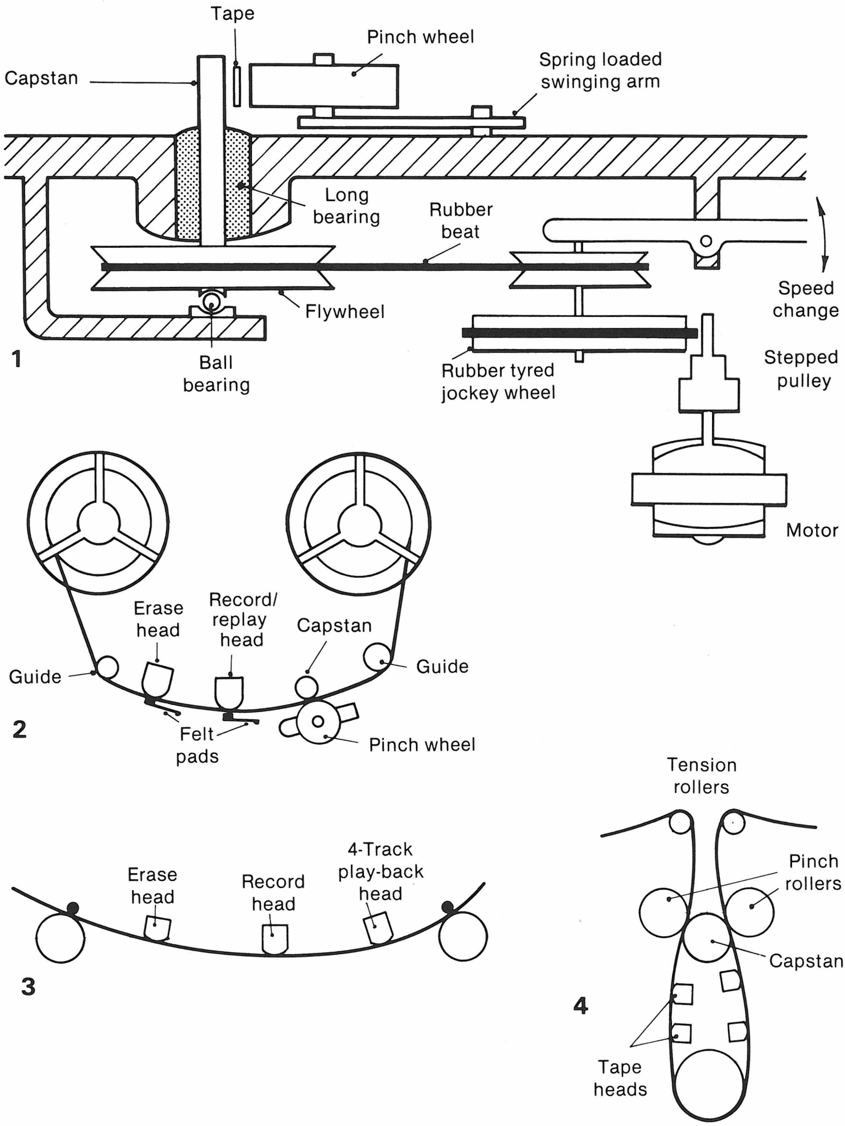
Domestic tape recorders use felt pads to press the tape against the heads, but in professional machines contact is maintained entirely by tape tension.

Reversible tape machines

A type of recorder has been developed in which the tape is driven on both sides of the heads. This is achieved either by using two sets of capstans and pinch rollers or by double pinch rollers acting on either side of a single large capstan as in the Technics tape recorders.

This arrangement helps to eliminate the problem of maintaining constant tape tension, particularly with machines without pressure pads and can result in very low wow and flutter.

Another advantage is the ability to reverse the direction of tape travel and thereby double the playing time. Reversible machines are usually fitted with two-track recording heads and four-track playback heads which are switched automatically when the machine reverses at the end of the reel.



1, Typical tape drive system for a simple recorder; 2, Typical tape path; 3, Twin capstan tape recorder; 4, Detail of an isolated loop tape recorder.

Tape Spooling Mechanisms

The tape is guided past the heads by polished tape guides which prevent tape movement laterally to the heads. They may be adjustable but usually it is the heads that are provided with adjustment facilities. By arranging the guides and heads in a slight curve, the tape can be kept in close contact with the heads as long as it is under tension. The tension is provided by applying slight braking to the supply spool. High quality machines rely entirely on this back tension for tape/head contact but domestic machines usually employ a felt pad to maintain contact with the record/replay head.

Take-up spool

The take-up spool is driven during record/replay either by its own motor or, in simpler machines, by the capstan motor through a slipping clutch. This is necessary to account for the change of speed as the diameter of the take-up spool increases and to prevent it from exerting too great a pull on the tape as applied to the pinch wheel.

Braking

As well as the slight friction braking of the supply spool during record and replay, a brake is required for instant stopping of the tape when the pinch wheel is withdrawn. The braking must be very efficient and should operate on both spools instantaneously, simultaneously and equally otherwise stopping the machine, particularly during fast spooling, would result in tape spillage or, worse, tape stretching.

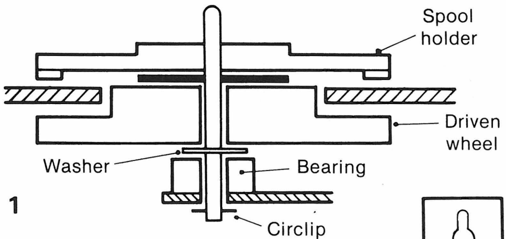
Tape position indicator

A digital counter is usually provided, with a reset facility, to enable a predetermined point on the tape to be found during spooling. In simple machines it is usually driven by a separate belt from the take-up spool. In high quality machines the counter is driven by a separate spindle over which the tape passes after the head block. In the more elaborate machines the counter is electronic. The spindle is usually ribbed around its circumference to prevent air being trapped between it and the tape which, during fast spooling, can cause 'lift off' and inaccurate reading.

Auto stop mechanisms

Most machines incorporate a lever in the tape path which, by pressing against the tape, detects its presence. When the tape runs out, the lever swings out and switches the machine off.

Other systems employ metal foil at the end of the tape to short out two contacts (usually incorporated in the tape guide) or a lamp and photoelectric cell arrangement. Either type of switch activates a relay which stops the machine or, in some cases, reverses it.



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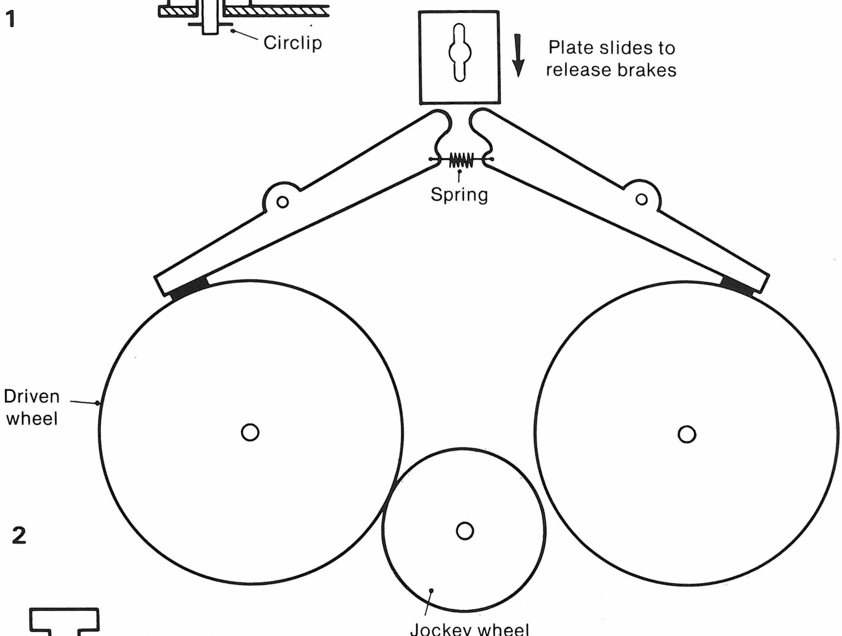

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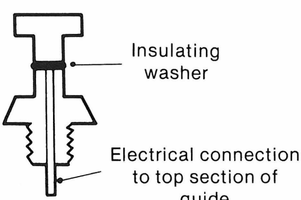

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1, Friction drive to spool holder; 2, Braking system; 3, Split (insulated) guide to activate auto-stop.

Tape Editing

One of the major advantages of reel-to-reel tape recorders, apart from their high quality capability, is the ease of editing the tape.

Cut editing

Accurate editing is achieved by physically cutting and joining the tape. By this means it is possible, with practice, to edit speech syllables or excerpts of music together unobtrusively without upsetting the rhythm. The method is also used for splicing on leader tape to enable accurate cueing when reproducing each sequence.

Cutting the tape

The procedure is to play back the tape until the cue is heard. The machine is then stopped and the two spools rotated slowly backwards and forwards by hand until the exact point of the cue is established. With practice it becomes possible to recognise cue points even at very slow speed. A mark is then made on the tape to coincide with the reproducing head (normally the one on the right) or against a marker post which is fitted to some machines at a measured distance from the reproducing head. The tape is then removed from the head and placed in a special guide, usually made of aluminium, which has a lip slightly narrower than the tape. The guide has a diagonal slit either at 45° or 60° to the tape. The mark on the tape is lined up with the slit or with a mark on the guide at the same distance from the slit as the marker post is from the repro head. The tape is cut by running a razor blade along the slit.

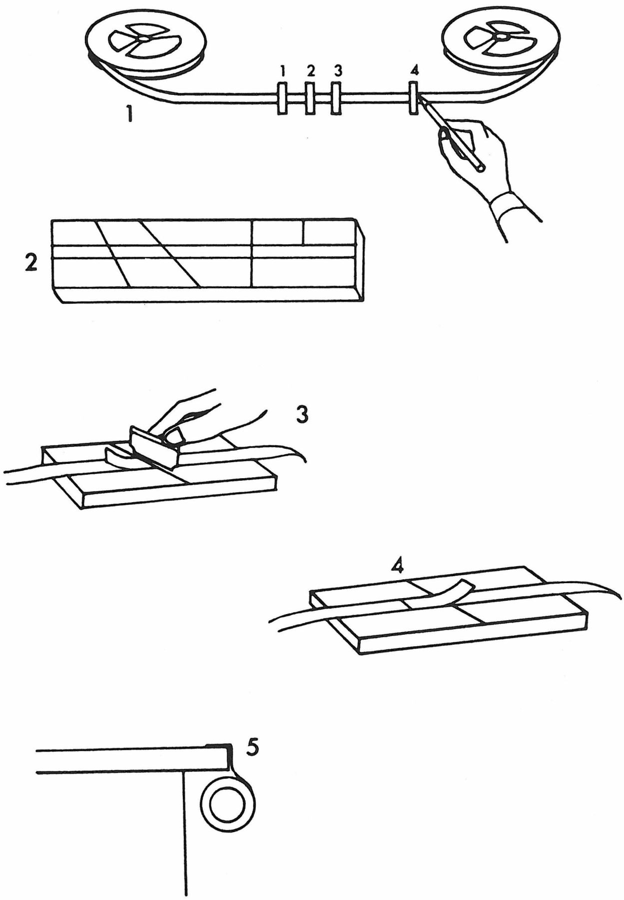
Joining the tape

The recording tape, or leader tape to which it is to be joined, is laid in the guide and cut in a similar manner. The reason for the diagonal cut is to create a smooth transition and prevent a click occurring due to a sudden change of signal level.

The two tapes are then butted together and about 3 cm of special adhesive tape is stuck over the join on the back of the tape. Care should be taken to put the adhesive tape on straight and to touch the sticky side as little as possible with the fingers.

Leader tapes

One convention is to use white leader tape (about 1 m) at the start and to write the title on it, short lengths (about 15 cm) of yellow leader between separate inserts, and red leader tape at the end. It is a good idea to get into a routine of marking the tape for splicing in a particular manner, eg always along the bottom of the tape. This avoids confusion and the risk of joining the tape on backwards when the splice is made.



1, Professional tape-machines have three heads: the erase (wipe) head on the left (1), record head in the centre (2) and playback head on the right (3). The editing point is found by 'rocking' the spools backward and forward. It is then marked with a yellow wax pencil either opposite the replay head gap or against a special 'marker post' (4).
2, An editing block with provision for cutting at 60°, 45° and 90°. When the block is used in association with a tape recorder with a marker post a mark is incorporated on the block at the same distance from the slit as the marker post is from the repro head.
3, The tape is cut with a non-magnetic blade together with the piece of tape to be joined.
4, About 1 in of joining tape is laid carefully across the join after removing the top cut off piece.
5, Operators usually stick the roll of the tape to the side of the machine for quick action. The area touching the table is cut off before use.

The cassette recorder combines the simplicity of disc with the flexibility of tape.

Cassette Recorders

The compact cassette recorder is effectively a miniature reel-to-reel tape system in which both spools are contained in a single plastic container thereby eliminating the cumbersome business of threading the tape. The cassette therefore combines the simplicity of operation and ease of storage of the gramophone (phonograph) record with the ability to record.

Mechanical arrangement

The normal domestic mains/battery machine uses a DC motor drive and an almost standardised drive assembly. A flywheel-driven capstan and pinch wheel transport the tape past the record/replay head with constant speed while the take-up spool is driven slightly faster than required through a slipping clutch. The motor and drive assembly are coupled via a combination of belts, pulleys and idler wheels. The change-over from record, replay, fast forward and rewind is completely mechanical; each pulley is mounted on a movable lever operated by mechanically linked push buttons, or via solenoids.

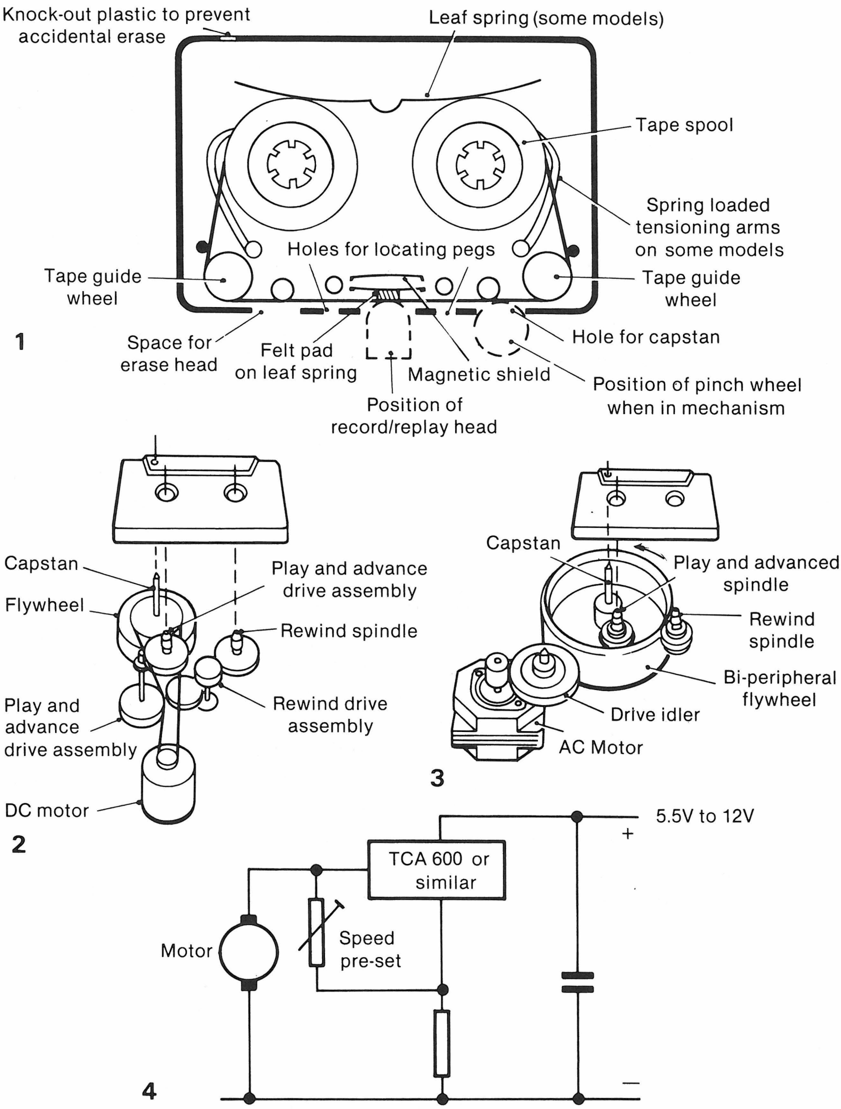
Problems of achieving high quality

Some of the limitations of this system are: low tape speed ($1\frac{7}{8}$ in/sec); narrow track width (the $\frac{1}{8}$ in tape has 4 tracks); and small spools.

The low tape speed and narrow tracks give rise to problems in head design, ie gap size, and the tracking arrangements required to ensure good azimuth alignment and head contact. The small size of the spools and the fact that the tape guides are contained within the cassette can result in wow and flutter which can vary from cassette to cassette and at different positions on the tape.

Near high fidelity results are achieved in some of the better models by various improvements in cassette drive mechanisms, which are usually powered by AC motors. One type uses a bi-peripheral flywheel to drive the spool spindles. Another uses a dual capstan-pinch wheel drive, ie before and after the record/replay head, in order to completely isolate the tape transport past the head from the changes in tension produced by the internal mechanism of the cassette.

Further improvement, even to the domestic machine, is also possible by introducing electronic speed regulation, the control system being contained in one integrated circuit. This can almost eliminate the speed variations due to such factors as varying supply voltage, ambient temperature variation, and variations of loading of the tape to the capstan (due to tape drag in the cassette).



1, View of cassette with top cover removed. There is a piece of knock-out plastic or hole covered by tape to prevent accidental erasure, which is removed for recording. The tape spool is free to locate on the spindle when inserted in mechanism; 2, A typical motor drive assembly; 3, Bi-peripheral flywheel drive assembly; 4, Electronic speed regulator. The control circuit is contained in a single integrated circuit, a TCA 600 or similar. The speed of the motor is adjusted by the variable resistance preset.

Cassette drive systems are arranged to locate and couple with the spools when the cassette is placed in the machine.

Cassette Drive Mechanisms

The drive to the tape reels, which are inside the cassette, is through a spindle assembly with spring-loaded dog clutches. These automatically locate into the slots in the reels when the rotation starts.

The right hand take-up spindle is driven through a slipping clutch to allow its speed to vary as the diameter of the tape on the reel increases. The tape speed past the heads remains constant.

Head assembly

The erase head, record/playback head, tape guides and pinch wheel are assembled on a sliding plate which moves forward in the record or replay position.

When the cassette is inserted the heads and pinch wheel penetrate it through the openings provided. This allows the heads to press the tape against the felt pad inside the cassette and the pinch wheel to press the tape against the capstan. The tape guides, which are attached to the head, hold the tape in correct alignment.

The pinch wheel is mounted on a swinging lever assembly, with a spring adjustment to give exactly the correct pressure against the capstan.

Drive mechanism

Except for the most expensive machines, a single motor drive is used. This is usually a DC motor. Some form of speed regulation is normal in all but the cheapest, most basic units. A single belt drive is used to the reels via pulleys and to the capstan. The combination of a somewhat elastic belt, rubber tyred pulleys and the large flywheel provides the necessary 'smoothing' required to maintain constant tape speed.

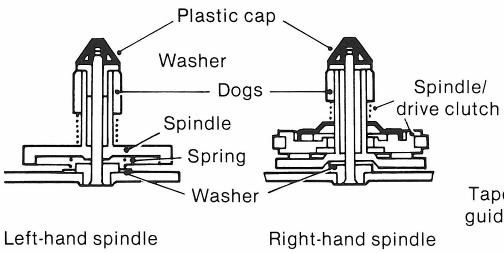
The different drive conditions required for record/replay, fast forward and fast rewind are provided by physical movement of the pulleys by a system of levers and sliding plates operated by push button controls.

In some high-quality machines, twin capstans are employed to improve the smoothness of the drive.

In the latest and most sophisticated machines, the use of logic control circuits, in conjunction with solenoid-operated transport mechanisms, enables the various functions to be selected by touch switches.

The logic system is programmed to operate in correct sequence so that if, for instance, the playback button is pressed during fast rewind, the machine will automatically come to a stop and change to playback in a smooth manner.

Some machines provide a 'search' facility whereby the machine can be 'programmed' to find a particular item on a recorded tape. It does this by counting the number of spaces between the items while in fast forward or rewind mode and comparing it with a preset figure.



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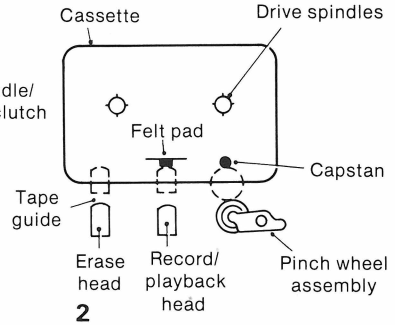

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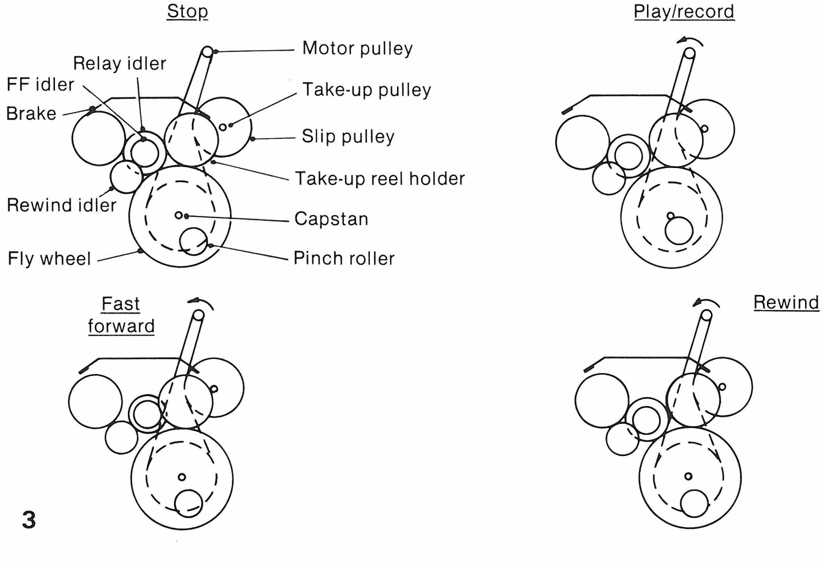

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1, Cross section of a left-hand and right-hand spindle; 2, The relationship between the cassette, the tape heads and pinch wheel. The pinch wheel, record/replay head and erase head move up to the dotted positions in record/replay modes; 3, Schematic diagram of typical tape drive mechanism showing the arrangement of pulleys and idler wheels for each of the four conditions: stop, play/record, fast forward and rewind.

Continuous development has resulted in cassette machines capable of quality comparable with disc.

High Quality Cassette Recorders

There has been continual development in the manufacture of tape, the electrical processing and in the design of the machines themselves, to the point where top quality cassette recorders are capable of providing remarkably good quality.

Double capstan drive

With the methods of drive described previously, the speed of the tape passing the heads is affected by uneven tension and snatch caused by variable torque in the take-up and supply spools. This is not acceptable for high quality machines and different methods of tape transport are employed.

In the double capstan system the tension on the belt between the motor pulley and the leading capstan is greater than the tension between the leading and the trailing pulley and between it and the motor. The tape between the two pulleys is thus kept taut, under slight tension, so that it passes over the heads, which are between the two capstans, at a steady speed.

Three head machines

When a single head is used for both recording and replay, it is necessary to make a number of rather unsatisfactory compromises. The ideal head gap size is larger for record than replay (see p. 90) but preference must be given to the more stringent requirements of replay. This means that it is difficult to achieve a satisfactory magnetic penetration when recording.

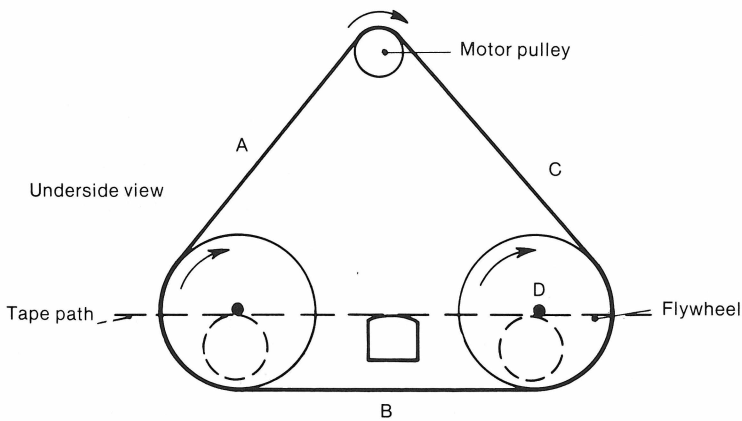
The situation is made worse if a ferrite head is used. This material is excellent for recording but can be rather noisy when used for replay with a very narrow gap, as it has a porous structure with random holes which can amount to a series of dummy gaps, producing random noise.

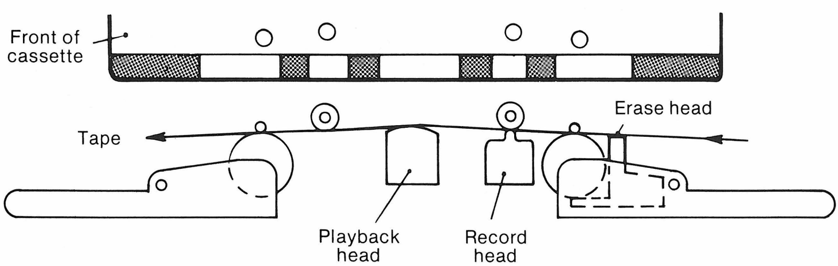
It is therefore preferable to employ ferrite heads for recording and erasing and mumetal heads, eg Permaloy, for playback. Another significant advantage of separate heads for record and playback is the ability to monitor a recording of the tape while it is being made.

Servo drive machines

It is very difficult to maintain constant tape speed with cassettes because of the varying friction of the pressure pad and imprecise guide rollers. Even slight variations will create wow and flutter and affect the frequency response by altering the tape tension.

These problems are overcome in high quality machines by controlling the tape speed by an electronic logic servo system driving the motor, controlled by a quartz-locked oscillator.





1, Underside view of cassette drive mechanism showing belt drive arrangement. The belt is taut at *A*, slacker at *B* and quite slack at *C*. The tape is therefore kept taut between the capstans (*D*); 2, General arrangement for a three-head twin-capstan machine, showing relationship of the heads to the cassette openings. High quality tape machines have separate record and replay heads so that each can have its optimum gap width—typically 5 micron for record and 0.7 micron for playback. It also enables simultaneous playback during recording for checking.

The eight-track cartridge system provides a convenient method of selecting between four stereo recordings or playing continuously.

The Eight-track Cartridge System

The tape of an eight-track cartridge system, like the cassette, is contained in a plastic box. Instead of two spools with tape feeding from one to the other, however, in the cartridge system a continuous loop of tape is wound on a single passive spool, feeding out from the centre and returning to the periphery. To attempt to pull the tape the other way would cause it to tighten on the centre boss and bind, which is why cartridge machines do not provide rewind facilities. Domestic cartridge players do not provide the facility to record.

The cartridge

The plastic cartridge contains a continuous length of $\frac{1}{4}$ in (6.25 mm) lubricated tape wound on a passive spool, suitable guide pulleys and a built-in pinch roller. A plastic sponge pad forms a backing to the tape opposite the slot into which the head penetrates when the cartridge is inserted into the machine.

The tracks

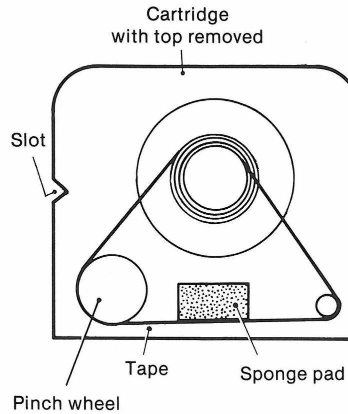
The tape is prerecorded using eight tracks in stereo pairs. Counting the tracks as numbers 1 to 8 from top to bottom the pairs are: 1 and 5, 2 and 6, 3 and 7 and 4 and 8.

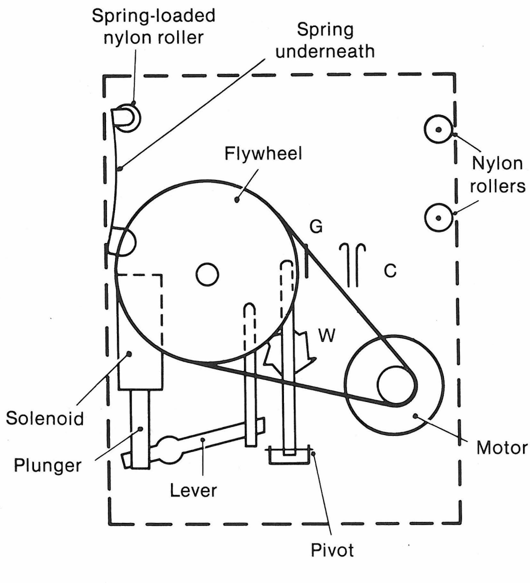
The head has two gaps which read these pairs to produce a stereo output. The head is moved vertically by a stepped ramp wheel so that, when it reaches the end of one pair of tracks (ie the loop has gone full circle), it steps on to the next pair, eventually stepping from bottom to top to start the cycle again.

The stepping mechanism is operated by a solenoid actuated by contacts which connect with a strip of metal foil at the end of each track. It can also be stepped by means of a push button to select the required track.

Tape drive mechanism

The motor drives a flywheel via a flat rubber belt. The flywheel spindle is the capstan and is supported by a substantial bearing. When the cartridge is slotted into position, the tape is compressed between the capstan and the pinch wheel, which is part of the cartridge assembly. Tension depends upon the pressure of the spring loaded nylon roller which sits in the triangular slot in the cartridge case. Head pressure is provided by a sponge pad within the cartridge. The position of the head relative to the tape is determined by the fixed tape guide. The head itself is pivoted in the vertical plane, its position being determined by the stepped ramp on the star wheel. This is rotated by the action of a lever, plunger and the solenoid which is activated by the metal strip on the tape passing the contact.





The eight track cartridge system. *G.* Fixed tape guide. *W.* Star wheel. *C.* Contacts to activate solenoid when shorted by metal section of tape.

NAB cartridge machines provide the facility of rapid selection and instantaneous cue.

Broadcast Cartridge Machines

Broadcast cartridge machines, employing the NAB format, are intended to provide a source of programme material that can be quickly selected and accurately cued. It is the ideal medium for inserting short recordings such as announcements, jingles, commercials and sound effects etc. Cartridges are sometimes used to record short sequences for post-dubbing to film.

Continuous loop cartridges

The cartridges contain a continuous loop of lubricated tape. This is pulled out from the middle of a single spool and returns to its periphery via guide pulleys, set in the corners of the plastic container.

Pressure pads are provided behind the tape at the points where the record and replay heads penetrate when the cartridge is engaged. There are three standard cartridge sizes, designated A, B and C. Spool lengths can vary from about 20 sec to 31 min, with the tape running at $3\frac{3}{4}$ in/sec.

Cueing system

Unlike the eight-track cartridge system which uses metal foil cueing, the NAB system uses audio frequency tones, recorded on a separate track to start and stop the machine and perform additional functions if required.

There are three standard cue tones in use: 1 kHz, 150 Hz and 8 kHz. The basic action uses the 1 kHz tone to stop the machine.

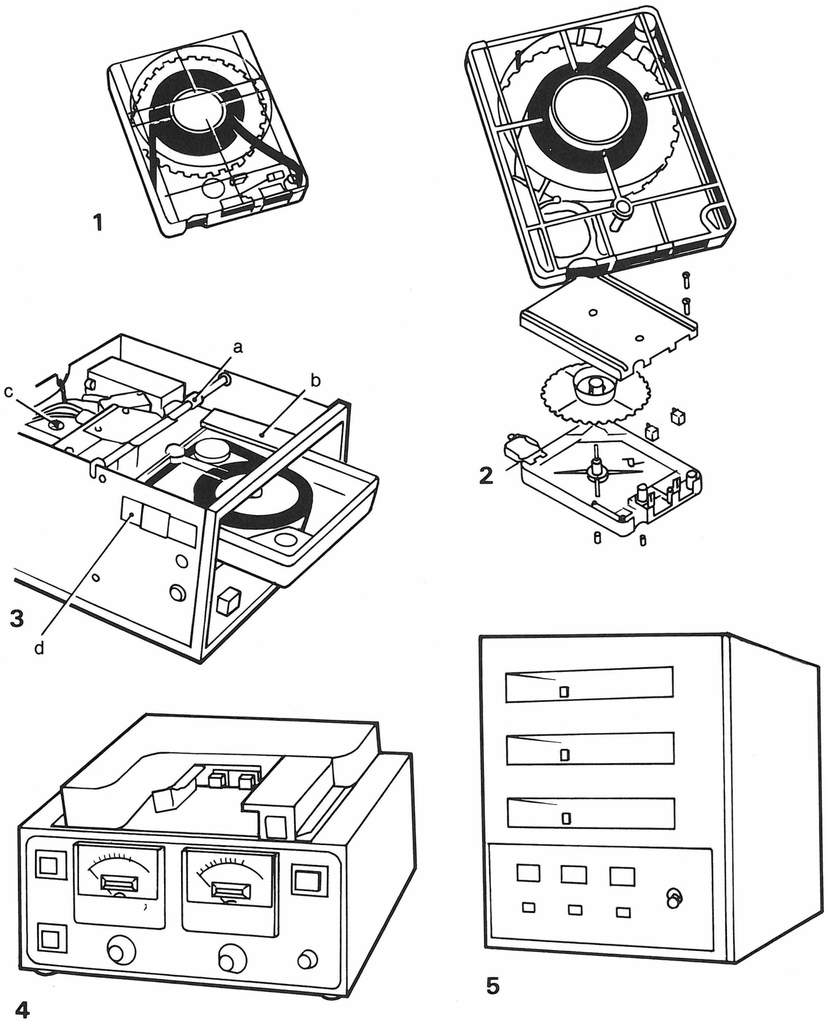
The action of putting the machine into the record mode records a short burst of 1 kHz tone on the cue track. When replaying the tape will stop automatically at this point. Thus if a 30 sec recording is made on a 40 sec cartridge, actuating the start button will cause the machine to play the 30 sec recording from the start. It will then run on a further 10 sec with no output until the cue comes round on the loop at 40 sec, where it will stop with the tape precisely located to play the cue again.

External cueing

Where machines are equipped with the two additional tones, the 150 Hz (auxiliary) tone is usually used as an end-of-message signal which can be used to start the next machine in a sequence. The 8 kHz (auxiliary) tone can be used for extra functions such as cueing superimposition of effects or picture slide changes etc.

Machine capability

Professional cartridge machines are capable of quite good quality. They are normally capable of mono or stereo recording and playback. Some have fast forward facility (two to three and a half times play) to expedite cueing. It is not normal to provide erase facilities.



1 and 2, Two types of cartridge. Note the difference in the position that the tape feeds from the spool; 3, Cartridge machine mechanism. *a.* Spring-loaded hold-down rollers. *b.* Tapered cartridge guide. *c.* Adjusting screws. *d.* Cartridge loading spring; 4, General view of cartridge machine showing position of the heads. This is a single machine; 5, Machines are available with up to three cartridge slots stacked in vertical formation.

Many cassette machines are battery driven and some form of speed control is needed.

Speed Control for Battery Cassette Recorders

Many cassette recorders employ DC motors so that they can be operated on batteries or as mains/battery units. When using a DC motor in a battery or battery/mains unit, the motor speed depends on the supply voltage which could vary with the life of the batteries or, for instance, vary from 12–15 volts if the supply is from a car. Some form of regulator is therefore necessary.

Simple mechanical governor

This is built into the motor and is a simple make-and-break contact, in series with the supply, operated by centrifugal force. The rapid fluctuations in speed are smoothed out by the usual flywheel and 'elastic' drive to the capstan.

The disadvantages are that it may become inefficient due to centrifuge spring fatigue and that it produces electrical interference which has to be suppressed.

Electronic governor

In the typical system shown opposite the speed is controlled by the current in the series translator TR1 which is biassed by TR2. The current through TR2 is fixed by the pre-set speed control but is also varied automatically.

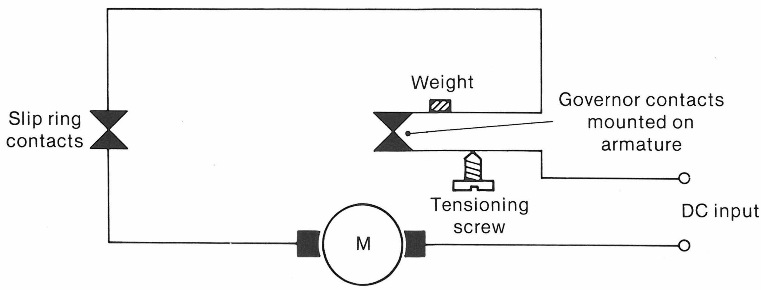
If the input voltage changes both the emitter voltage (via R1, D1 and D2) and the base voltage (via R4, R5 and R6) will change but it will always be smaller on the base. The change in current through TR2 and subsequently through TR1 maintains the correct motor current and speed. Any change in load alters the motor speed and thus the back emf; the motor current and the potential difference across R2, therefore, also change. This is effectively a change of bias on TR2 which again re-adjusts the motor current via TR1.

The resistor R, provides the necessary starting current as TR1 and TR2 will be off until the motor begins to turn.

AC feedback governor

This is a more elaborate system which depends on the output from an AC generator mounted on the motor armature. The frequency depends on motor speed; by first integrating and then rectifying a DC control voltage is produced. This is used to bias a transistor in series with the DC motor.

In better quality machines, AC motors driven by quartz-locked oscillators and power amplifiers are sometimes used.



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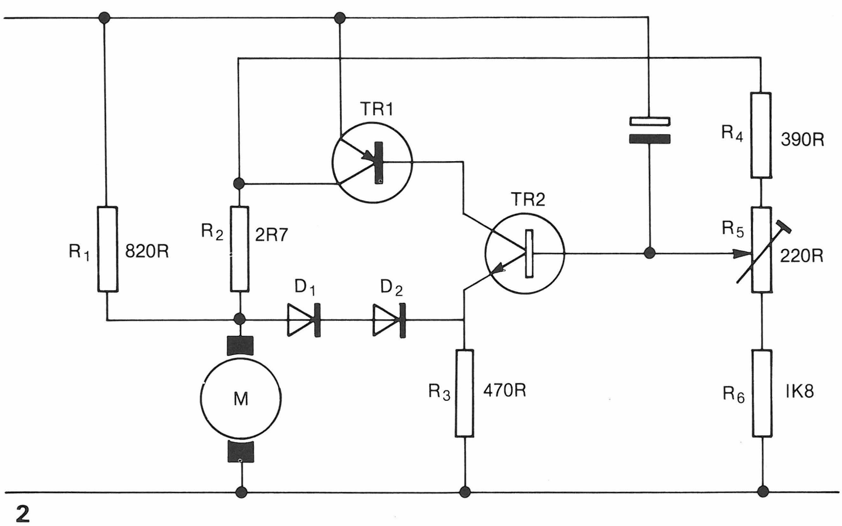


2

1, Mechanical governor. The armature rotates and, at a speed determined by the weight and the screw, the governor contacts open. This causes a reduction in speed when the contacts close again. The speed settles to the value at which the contacts are continually opening and closing. The sporadic nature of the drive is cancelled out by the flywheel pulley; 2, Typical circuit for an electronic governor.

Good quality sound output is completely dependent on constant head speed and a close contact between tape and head gap.

Tape Recorder Faults and Mechanical Adjustments

Most mechanical faults cause a variation of sound output pitch which is caused by the fluctuations of tape speed.

The slow variation of speed called 'wow' and the quicker variation 'flutter', are most easily detected by listening to a 3 kHz tone from a standard test tape. Wow is introduced by a fault associated with a rotating part of the mechanism; the wow frequqnecy will be the same as the frequency of rotation or a multiple of that frequency. This means that the faulty part may be spotted visually by comparing the speed of rotation with the frequency of the wow. Flutter is more likely to be introduced by parts of the mechanism which exert friction on either the tape or the spools. The fault can usually be traced by slightly easing each friction point in turn.

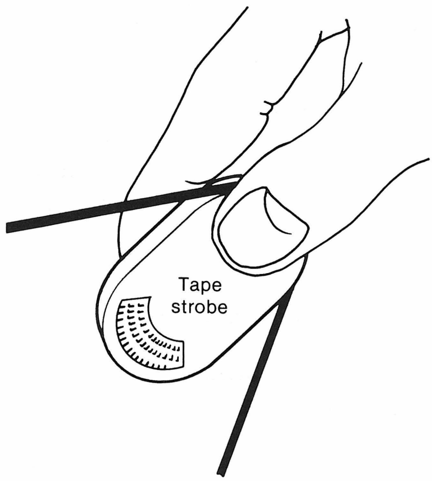
Incorrect speed and uneven running generally result from more severe examples of the type of fault that causes wow and flutter and are therefore relatively easy to trace by the same means.

Mechanical causes of poor sound quality usually concern poor tape-to-head contact or dirty or worn heads.

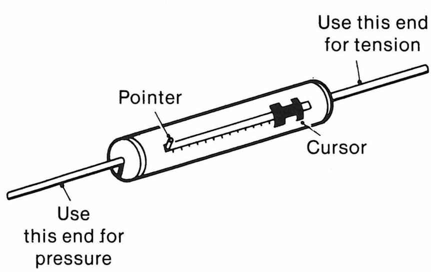
Fault tracing

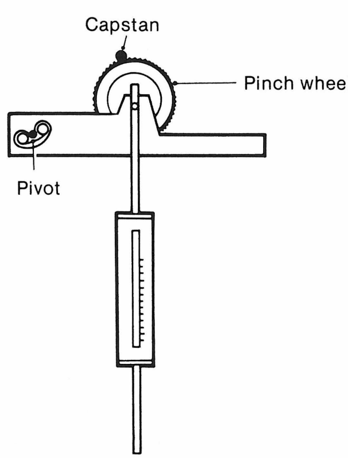
Fault condition	*Possible causes*
Wow	Slack drive belt. Loose flywheel. Slightly worn bearings. Incorrect pinch wheel tension Oil or grease on pinch wheel. Warped spools.
Flutter	Sticky pads or brakes. Sticky tape pressure pad. Slipping clutch with uneven surface. Oxide deposit on guides or capstan. Too much tension. Deformation of pinch wheel through being left engaged while stationary.
Uneven running/ slow speed	Very slack drive belt or jockey pulleys. Twisted or kinked belt. Sticky or badly worn bearings. Sticky clutches or brakes, possibly impregnated with dirt or oil.
Poor quality (often coupled with low output)	Dirty or worn heads. Incorrect pressure pad tension. Incorrect back tension (where pressure pad is not used). Dirty switch or relay contacts. Tape fitted back to front.

A portable strobe to test speed. The tape is looped around the wheel causing the strobe striations to rotate. The striations indicate speeds of $7\frac{1}{2}$, $3\frac{3}{4}$, $1\frac{7}{8}$ and $\frac{15}{16}$ i/s. When viewed under a 50 Hz light source, one band should be stationary indicating the correct speed.



Pressure/tension gauge (spring-balance type) suitable for checking pinch wheel pressure. The gauge has a rod at one end to measure pressure and a hook at the other to measure tension. The method is to set the cursor to zero and then press or pull against the pressure or tension being measured. The cursor remains at the maximum reading obtained.





Using the tension-testing end of the gauge to measure pinch-wheel pressure.

Tape Recorder Amplifiers

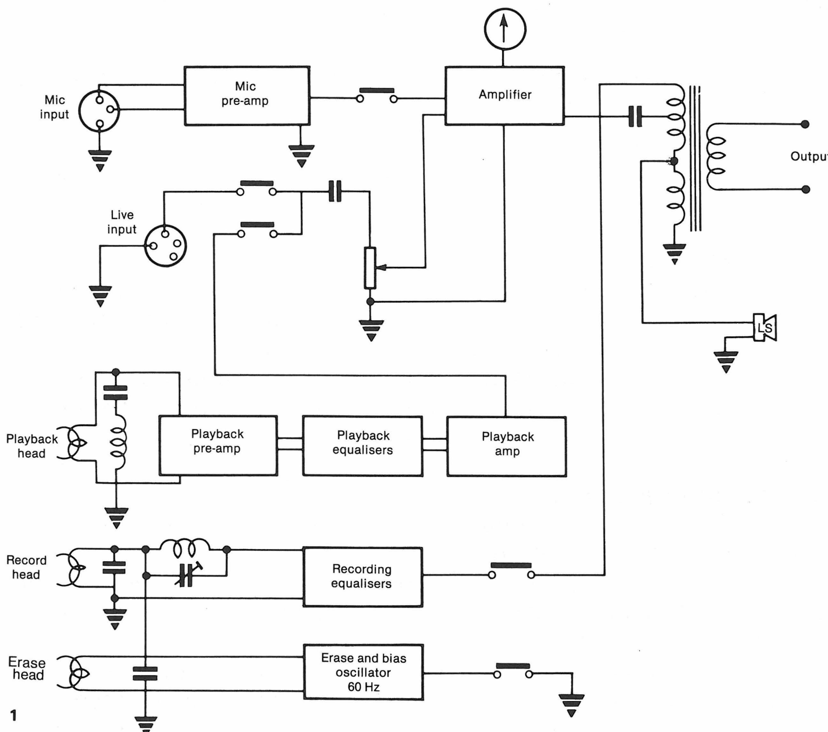
Tape recorders require amplifiers to drive the recording head or to magnify the signal from the replay head and to apply the appropriate equalisation.

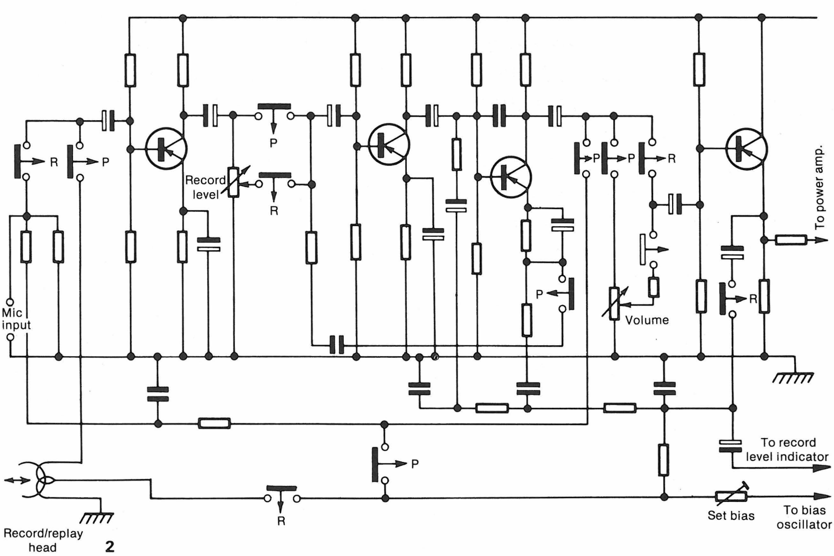
High quality tape recorders
High quality tape recorders employ separate record and replay amplifiers connected to separate record and replay heads. This simplifies the problem of obtaining the correct response for each function including the provision of different equalisation for different tape speeds, types of tape and recording standards, eg NAB, CCIR etc (see p. 110). It also makes it possible to monitor from the replay head while a recording is being made.

Amplifier requirements
Recording amplifiers must be capable of supplying the record head with a signal current of about 0.5 mA, possibly from a microphone input varying from about 0.0001 volts (ribbon microphone) to about 0.005 volts from a crystal microphone, pick-up or radio receiver etc. The playback amplifier has to cope with an output from the playback head that rises from about 0.00005 volts at 50 Hz to 0.001 volts at 1 kHz and tails off again in the high frequencies (see p. 104).

Domestic tape recorders
In the cheaper domestic machines, where a common record/playback head is used, the amplifier, except for the power stage, is usually made common to both record and playback by the use of some rather complex switching. In the simplified circuit shown opposite, which may be taken as typical of an average cassette recorder, all the switches shown are ganged, ie they operate together. *R* represents closed on record; *P* represents closed on playback.

1, Block schematic diagram of basic amplifier arrangement for a three-head tape recorder; 2, Simplified circuit diagram of typical cassette recorder amplifier.


Mic
input
Mic
pre-amp
Amplifier
Output
Live
input
LS
Playback
head
Playback
pre-amp
Playback
equalisers
Playback
amp
Record
head
Recording
equalisers
Erase
head
Erase and bias
oscillator
60 Hz
1


R
P
P
Record
level
R
P
P
R
To power amp.
Mic
input
P
Volume
R
P
To record
level indicator
Record/replay
head
R
Set bias
To bias
oscillator
2

Some adjustments affect both record and playback while others affect only one or the other. It is recommended that the playback adjustment should be done first.

Tape Recorder Electrical Adjustment

Playback adjustment

Position of head. If the head position is incorrect, there may be poor high frequency response, low volume or cross-talk between tracks. Adjustments should be made with a signal from a test tape, eg a 5 kHz tone. An EVM across the loudspeaker is a suitable indicator. Head position is adjusted for maximum output. With a stereo machine, optimum performance is obtained by adjusting the head for maximum M signal (A+B) and minimum S signal (A−B).
Equalisation. If the equalisation is incorrect the frequency response will fall short of specification. Adjustment should be made with a 15 kHz test signal for maximum output.

Recording adjustment

Bias oscillator. It is essential that the frequency and output level should be correct for good quality recordings.
Frequency. Frequency should be checked with an oscilloscope and set to the frequency specified for the particular machine. A typical frequency is 100 kHz.

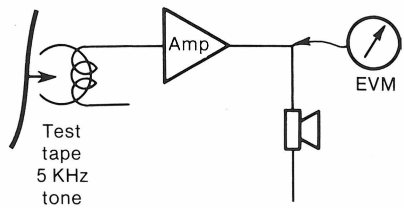
Note that some machines have a switch facility for altering the frequency to avoid beating with 38 kHz sub-carrier when recording from a stereo radio transmission. The resultant loss in efficiency is preferable to a constant audible tone.
Bias current. The bias current to the record head is usually measured by checking the voltage drop across a resistor in series with the head. The 'bias level control' is adjusted to give the specified voltage. It is usual, however, to produce a series of test recordings at spot frequencies between 80 Hz and 10 kHz. The output levels, at full volume, should have less than a 6 dB variation. If not, the bias level should be re-adjusted.

Too much bias current will reduce the treble output.

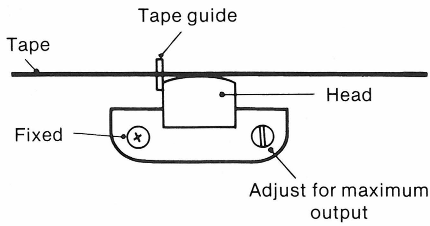
Too little bias current will give distortion at high modulation peaks. See p. 98 for details of optimum bias setting.

Other variations of frequency response will require adjustment to the record equalisation network in the amplifier.
Record level indicator. If a series of test recordings is produced at progressively increasing input levels, the level at which distortion occurs can be determined on playback. Adjust the record level pre-set control so that the indicator overloads at that input level.
Tape speed control. Adjust tape speed with a test tape of specified length, eg $93\frac{3}{4}$ in (100 sec at $\frac{15}{16}$ in/sec). The speed should be correct within ±5% (±5 sec). If available, a stroboscope can be used. Test tapes are available with long 'leader' sections made of alternate black and white segments to act as a stroboscope.

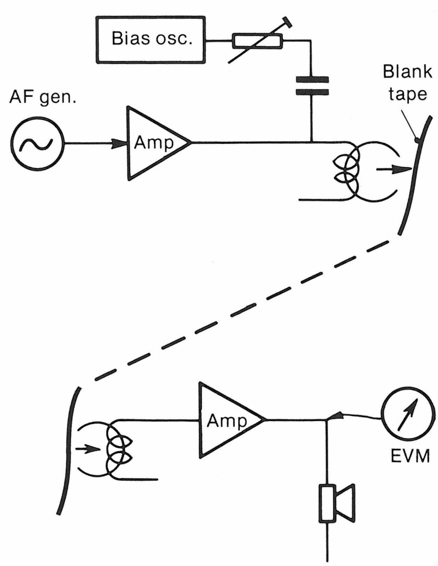
Playback adjustments.



Recording adjustments.
Record at, typically, 300 Hz and 6 kHz.
Playback and compare levels.



Adjust the bias level until levels are within, typically, ± 2 dB.



High quality disc reproduction begins with a good turntable.

Disc Reproduction

The range of equipment now available for the playing of disc recordings offers a wide choice of arrangements from the complete compendium of the 'music centre' to the 'hi-fi' arrangements of independent units each chosen for a specific function.

Disc player arrangements

The popular disc player usually comprises a mains powered motor driven turntable with three or four speed selections, a tracking arm with a wired-in turn-over cartridge, an integral amplifier with volume and tone controls together with a loudspeaker unit contained within a single cabinet. Usually there is provision for connecting an external loudspeaker if desired.

In contrast, a hi-fi installation could consist of a transcription deck with single or multi-speed selection, with the turntable incorporating a stroboscope for use with a fine speed control; an adjustable tracking arm allowing a choice of pickup cartirdges the output of which might be fed to a separate pre-amplifier unit with selectable pick-up compensation; tone controls and switchable rumble and hiss filters. This would then feed a level-controlled main amplifier capable of the high output level needed to drive an acoustically tailored loudspeaker system. The arrangement should be capable of accepting mono, stereo or quadraphonic discs.

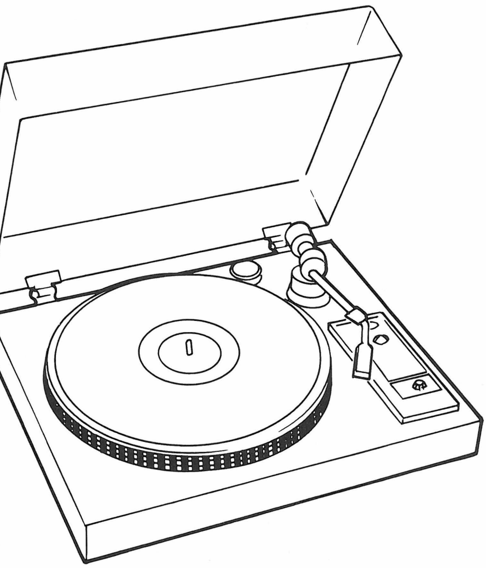
The stroboscope

The stroboscope is a device for checking turntable speed. It can be purchased or made in the form of a disc marked with a pattern of radial black and white bars which is placed on the turntable. Modern high quality decks have a suitable pattern printed around the rim of the turntable, and some (usually older types) have a pattern of holes incised through the rim for the same purpose.

When viewed with a mains-operated lamp (preferably of the neon type), run from a mains AC supply, the pattern for the speed chosen will appear to be static only when the speed directly relates to the mains frequency. The number of bars in the pattern therefore depends on the required speed and the mains supply frequency, as listed in the table opposite.

These values derive from the relationship:

$$\text{Number of alternate bars} = \frac{2 \times \text{Light flash frequency (Hz)}}{\text{Turntable speed (revolutions/sec)}}$$



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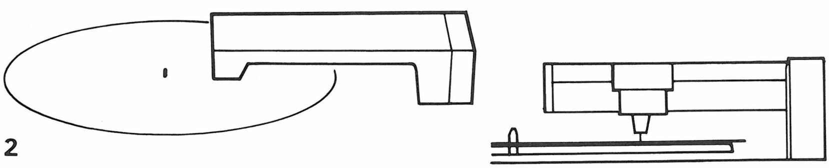


2

NUMBER OF BLACK AND WHITE BARS IN STROBOSCOPE PATTERN

Mains frequency	*Turntable speed* (revolutions per minute) $16\frac{2}{3}$	$33\frac{1}{3}$	45	78
50 Hz	360	180	133	77
60 Hz	432	216	160	92

These values derive from the relationship:

$$\text{Number of alternate bars} = \frac{2 \times \text{Light flash frequency (Hz)}}{\text{Turntable speed (revolutions/sec)}}$$

1, Typical arrangement for a high quality transcription turntable and pick-up. Having the disc player as a separate unit enables it to be placed well away from the loudspeakers thereby reducing the risk of acoustic feedback. 2, An alternative method of tracking, using a parallel tracking mechanism. The pickup is moved along the track by an electronically controlled servo mechnism. With some examples of this arrangement the stylus to record pressure is not affected by gravity so that the turntable need not be horizontal and can in fact be operated vertically.

High quality reproduction demands a constant speed mechanism.

Turntable Drive Systems

The basic requirements for good sound reproduction are speed, accuracy and stability.

The motor unit

Turntable motor units are usually of a type similar to the capstan-drive motor in a tape recorder (see p. 114). These units can be: synchronous motors, in which the speed of rotation is locked to the frequency of the mains; hysteresis-synchronous, which incorporate a hard-steel shell in the rotor to provide a measure of permanent magnetism and therefore stronger 'lock-up'; or, brushless DC motors with servo control. Synchronous motors can be driven directly from a stable oscillator through a power amplifier. The motor, or turntable platter, can be fitted with a strobe tachometer, which can be a light and photo electric cell or electromagnetic sensor. This produces a reference frequency which is compared with that of a stable oscillator and any difference signal is used to make the necessary speed correction.

Turntable drive

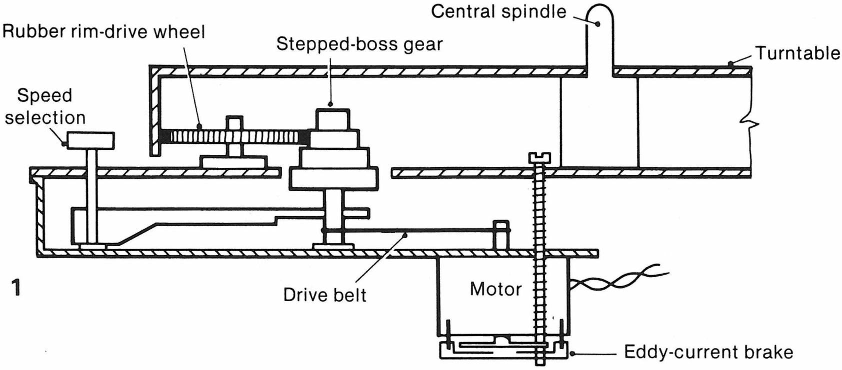
There are three methods of driving the turntable platter in common use: rim drive, belt drive and direct drive. In the rim drive method, a stepped boss transmits a friction drive via a rubber-rimmed idler wheel to the inside rim of the platter. Speed change is effected by raising or lowering the boss, or the idler, so that the idler engages with a different diameter on the boss. A mechanical linkage disengages the idler wheel when the motor is switched off to prevent the development of 'flats' which would cause flutter.

In the belt drive method, the turntable is driven by a flexible belt stretched between a large flywheel under the platter and a stepped pulley driven by the motor. The speed can be changed by raising or lowering a 'fork' which causes the belt to engage with a different diameter section of the pulley.

In direct drive, a multi-pole slow-speed motor drives the platter spindle directly. Speed can be changed by altering the frequency of the AC supply or switching the number of poles in the motor.

Rumble

An important parameter in the choice of turntables is freedom from rumble, ie low frequency noise transmitted from the motor to the pick-up, through either the platter or the tracking arm pivot. Stereo pickups are sensitive to vertical vibration. The level of rumble should be at least 45 dB below the signal level. This can be achieved with well designed and independently suspended motors and drive systems.



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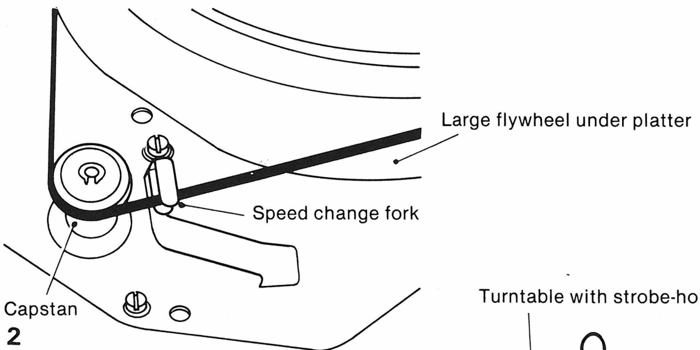

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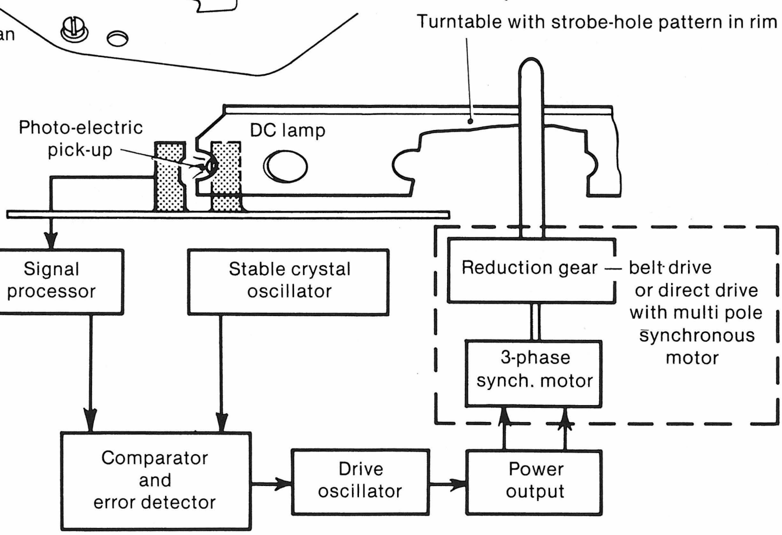


3

1, Outline of a four-speed deck; 2, Belt-driven turntable arrangement. The speed change fork moves the belt to different diameters on the capstan; 3, Block outline diagram of self-controlled power oscillator system.

Speed stability and high starting torque can be achieved by a direct drive motor controlled by a quartz oscillator.

Crystal Drive Turntable

The search for speed, accuracy and stability in turntable drive systems has led to the introduction of the crystal-controlled drive system pioneered by Technics. The basis of the system is a direct-drive turntable driven by a DC motor with servo control.

Speed control

The speed of the DC motor is locked to an oscillator controlled by a quartz crystal. The oscillator produces a very stable reference frequency which is split by a frequency divider into an appropriate control frequency for the selected turntable speed ($33\frac{1}{3}$, 45 or 78.26 rpm).

Integral with the drive motor is a frequency generator which produces an AC signal related in frequency to the actual turntable speed. This is read by speed and phase control circuits. The speed control circuit converts the output of the frequency generator into the appropriate electrical voltage required to maintain correct speed regardless of changing load conditions.

The phase control circuit matches the phase of the signal derived from the frequency generator to the phase of the divided-down reference signal. Differences in phase are translated into current to the servo motor in either a forward or reverse direction as required to maintain correct speed. The phase-locking action provides very fast correction torque which, in conjunction with a relatively heavy platter, produces very accurate speed stability and almost complete freedom from wow or flutter. The direct drive and heavy platter, coupled with vibration damping, result in very low figures for rumble.

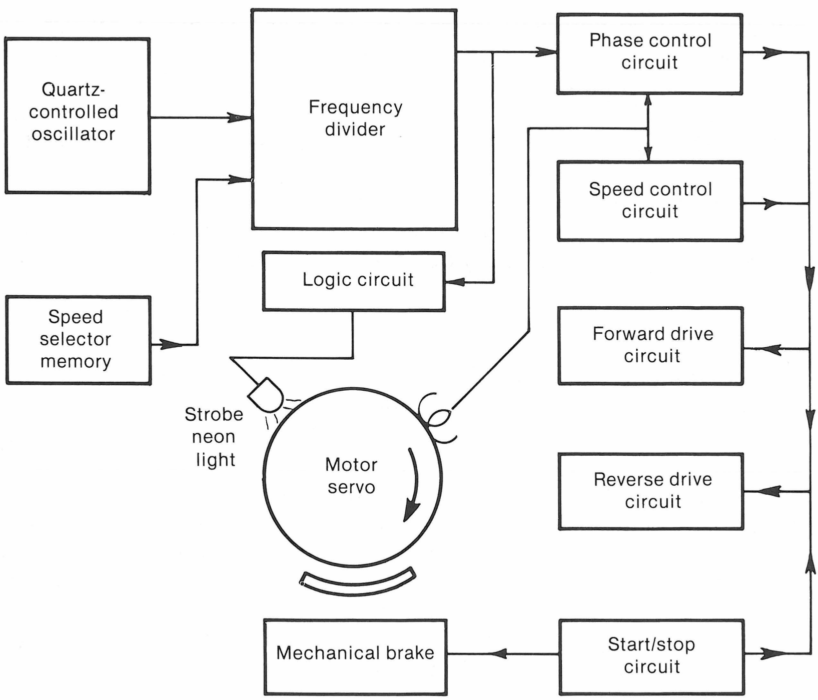
This type of turntable has an exceptionally fast start, attaining the full speed of $33\frac{1}{3}$ rpm in approximately 25 degrees of revolution. It also has an electromagnetic and mechanical breaking system which stops it almost instantaneously.

Stroboscope

A strobe pattern is included on the periphery of the platter which is viewed with the light of a neon lamp fed (via special waveform-shaping circuitry) from the divided-down reference frequency. It is thus independent of mains variations and requires only one strobe pattern to accommodate the different speeds.

Pitch control

The majority of turntables provide a fine adjustment of speed within a range of about 10%. In better quality quartz-controlled systems this is achieved by altering the frequency of the oscillator.



Block diagram of the Technics SP-10 Mk II turntable drive system.
The frequency divider divides the oscillator frequency down to a suitable control frequency for the selected speed. The speed selector memory stores the selected speed and dictates the appropriate division to the frequency divider. The phase control and speed control circuits compare the frequency and phase of the divided control frequency with that derived from the frequency generator on the DC motor (FG). The resulting control signals are interpreted by the forward and reverse drive circuits and applied to the motor servo. A strobe neon lamp has a waveform specially shaped by the logic circuit of the divided reference frequency. Mechanical and electrical braking is applied by the start/stop circuit and the mechanical brake.

Reproducing Styli

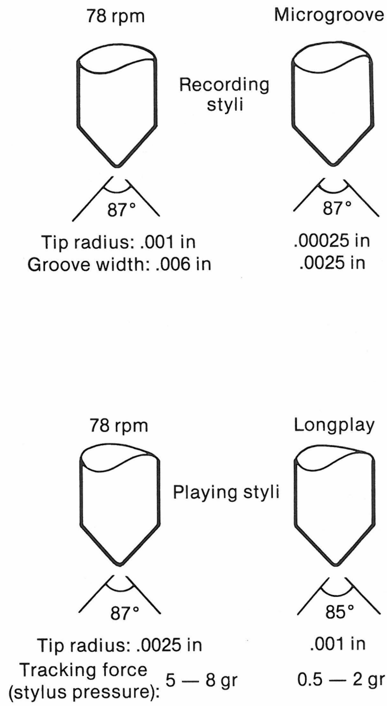
The excessive weight of early pick-up heads and the rapid wear suffered by steel or 'thorn' styli caused serious groove wear. The life expectancy of the early 78 rpm disc was very short.

Stylus shape
In order to obtain more faithful reproduction, modern pick-up styli are designed to complement the shape of the recorded groove. The shoulders of the stylus must track the sides of the groove, and the point must penetrate but not reach the bottom of the valley. To achieve this, the radius of the point is made about four times greater than that of the recording stylus. Enough tracking force must be applied to maintain full contact with the groove walls at all times whilst at the same time cause only negligible wear. It must be hard enough to stand up to the mechanical forces to which it is subjected during use and be ground to a very smooth finish to keep contact friction low.

Stylus materials
Various materials are used for modern styli and are usually of a hard crystalline material. High quality steel is no longer common, sapphire is most popular with ruby or diamond being used in the more expensive cartridges. Although of hard materials, styli are liable to be chipped by rough handling and can then cause severe damage to the soft material of the modern microgroove disc. They have long working lives, ranging from several hundreds of playings for sapphire to several thousands for diamond before noticeable wear. The amount of material used in most modern cartridge styli does not allow for regrinding but styli in the form of a tipped shaft or needle can be reground for further use.

Trailing angle
The recorded groove is cut with a vertical stylus but, when replayed, optimum tracking occurs if the pick-up stylus trails at slight angle from the vertical, usually about 10 to 15 degrees. If the stylus is of circular cross-section this effectively increases its diameter longitudinal to the groove and results in some high frequency attentuation. This can be overcome by giving the stylus an elliptical cross-section at the point of contact making its longitudinal thickness approximately proportional to the sine of the trailing angle. The tracking achieved with the elliptical stylus is very accurate.

High frequency attentuation due to tracking error is progressive towards the centre of the disc but this is compensated for during recording and does not require correction.



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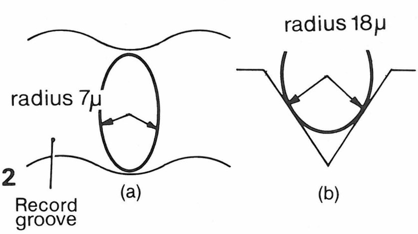

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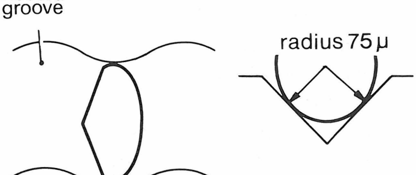


3

1, Comparison of recording and replay styli. 2, The replay stylus with trailing angle 10 to 15 degrees and elliptical stylus with longitudinal cross-section elliptical (*A*) and contact cross-section circular (*B*).

Stylus Tracking

The modern microgroove disc requires a very high standard of performance by the pickup head and this depends very largely on the ability of the stylus to follow the groove variations.

Trackability

To obtain a good high frequency response with minimum wear to the stylus or record groove the tip mass and tracking weight must be as small as possible and the compliance (ie the ease with which the stylus can be moved from side to side) as high as possible.

On the other hand the stylus weight must be sufficient to enable it to stay firmly in the groove while being subjected to the very rapid accelerations involved in reproducing high frequencies and transients.

Tracking distortion

Any failure of the stylus to follow the contours of the groove will result in either deformation of the grooves (possibly resulting in permanent damage) or cause the stylus to ride up and in severe cases skate across the record. In either case tracking distortion will occur which produces a 'roughness' in the character of the sound due to the addition of spurious harmonics.

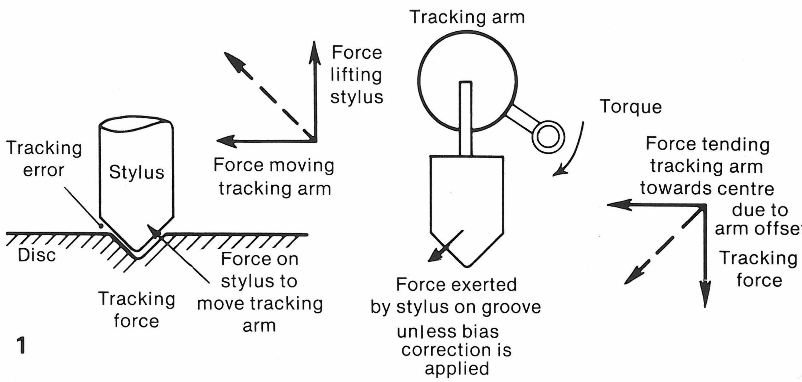
Bias compensation

A possible cause of tracking distortion is the mechanical bias imparted to the pickup due to the angle of the arm with respect to the groove. Because the pickup arm is not directly in line with the groove, the drag of the stylus in the groove creates a force which acts to one side, tending to pull the arm towards the centre of the record.

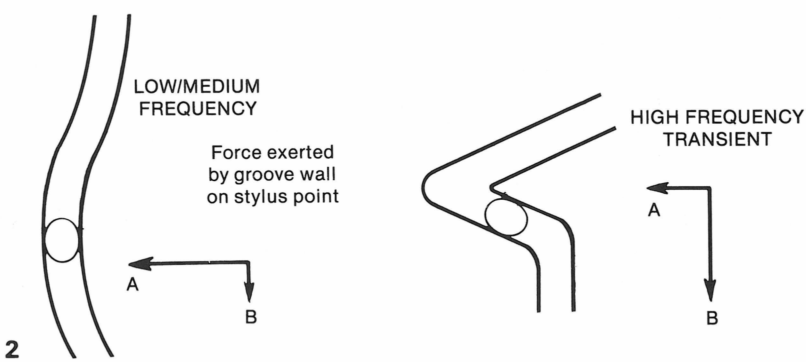
To compensate for this most tracking arms are provided with 'Bias Compensation' (sometimes called 'Antiskate'). This usually takes the form of a weight suspended on a string, a magnetic bias or specially angled tension springs. The sideways 'bias' is made adjustable and should be increased in relation to the tracking weight.

Bias can be assessed using a test record with tones of various frequencies recorded at very high levels and adjusted for the minimum required to prevent audible harmonic distortion (impurity of tone). Another type of test record has bursts of filtered tone which are adjusted for visual symmetry with the aid of an oscilliscope. The well-known Shure 'Audio Obstacle Course ERA 111' provides a test using musical excerpts, recorded at five increasing levels, for the subjective assessment of trackability under dynamic conditions.

Trackability also involves the ability of the pickup to ride the up and down motion of a warped disc. This is a function of the low frequency resonance of the pickup/tracking-arm combination.



1, Tracking error. Unless dynamically balanced, part of the groove force is used to move the tracking arm which can result in tracking error.
With the tracking arm dynamically balanced, the groove 'guides' the arm towards the disc centre.



1, Comparison of Recording and Replay Styli; 2, The replay stylus with trailing angle 10 to 15 degrees and eliptical stylus with longitudinal cross-section eliptical (*A*) and contact cross-section circular (*B*).

The Tracking Arm

The purpose of the tracking arm (or tone arm) is to support the pick-up head as it tracks the groove across the disc. Two basic types are in use: the radial arm and the pivot arm.

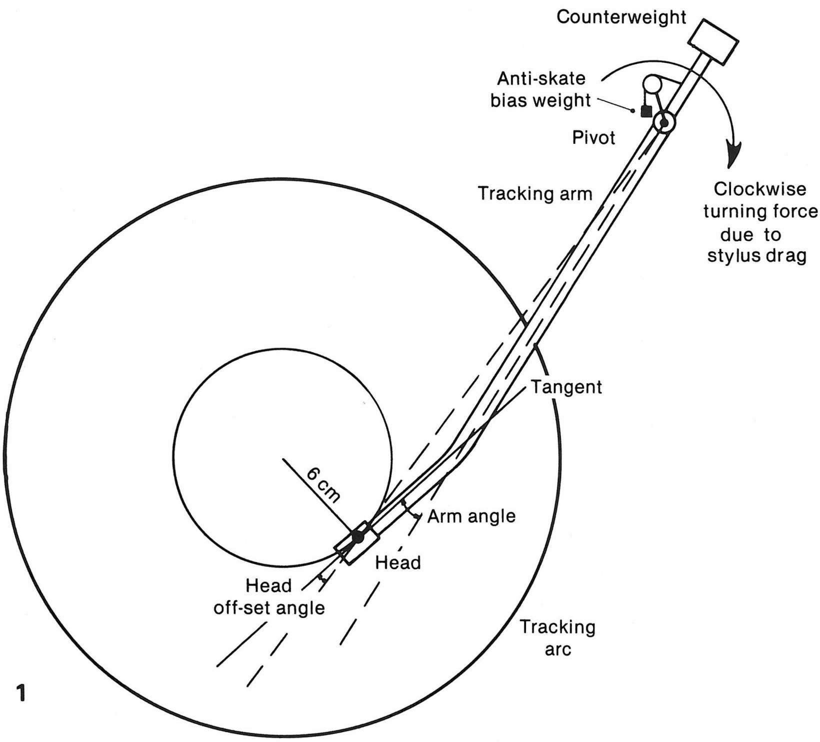
The radial arm
The arm and head form a straight line and track linearly so that the head is always tangential to the recorded groove. This would appear to be an ideal arrangement as the action is similar to that of the cutter when the groove was cut but design problems make this type of arm expensive.

The pivot arm
As its name implies, the arm pivots about its fulcrum and thus the head travels in a shallow arc across the disc. Tracking errors are unavoidable but are by no means as serious as first appear. In fact, the compromises in design are small and arms capable of the highest performance can be economically manufactured.

To improve the tracking, the head is off-set from the radius of the pivot to be tangential to the groove at an optimum point, which is empirically determined to be 6 cm from the centre of the disc. The arm and head are balanced horizontally about the pivot and then loaded to give the required stylus pressure in the groove, either by gravity (off-balancing) or by spring loading. The action of the fulcrum and pivot must have minimum resistance.

Arm resonance
The mechanical interconnection of the groove, stylus, head, arm and pivot form a group of resistances, compliances and masses which equate to resonances and impedances referred to the stylus point. The impedances tend to interfere with the tracking action of the stylus causing distortion of the signal being reproduced. The resonances give rise to unwanted signals. Other external effects include surface ripple on the disc, misplacement of disc centre and turntable vibration (rumble) due to the drive system or motor, cabinet, floor etc. A judicious application of choice of materials, balancing and damping during the design and construction can reduce these effects to a minimum. Resonances are constrained wherever possible to occur outside the range of the frequency band being used.

Arm–head relationship
Generally, tracking arms are designed for use either with an integral head or with replaceable heads from the same manufacturer. Some arms are designed for general purpose use with adjustable fulcrum, pivot, balance and torque.



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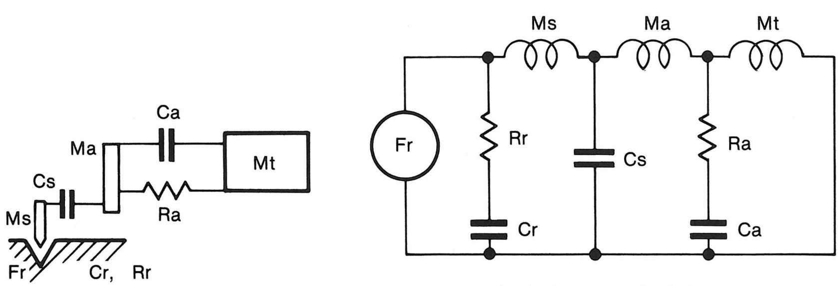

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1, With the optimum tracking point located 6 cm from centre of the disc, the effect of the arm angle is to reduce the error from the tangent along the tracking arc. Off-setting the head introduces torque at the stylus point which needs counter-balancing; 2, Basic head-arm circuit and equivalent electrical circuit. *Ms*=stylus mass; *Cr*= groove compliance; *Rr*=groove resistance; *Cs*=stylus–transducer compliance; *Ma*= transducer mass; *Ca*=mounting compliance; *Ra*=mounting resistance; *Mt*=tone arm mass.

Monophonic Pick-up Heads: Magnetic

The purpose of the pick-up head is to convert the fluctuations in the recorded groove into an electrical signal which can be processed and converted into as close a copy as possible of the original sound.

The magnetic pick-up

The principle of operation is the conversion of vibrations induced in the stylus in tracking the groove into variations in a magnetic circuit which in turn cause the generation of a voltage in a coil of wire. The voltage induced in the coil is proportional to the rate of change of the disturbance. Thus the output voltage will have a characteristic which rises with frequency at a rate of 6 dB per octave.

In order to keep the total mass small, the magnets and windings have to be small, resulting in a low output impedance and low signal levels. The output of such heads will therefore need to be carefully matched and compensated for electrically.

These heads are also susceptible to induction from nearby magnetic fields, such as from the drive motor, and so require careful design and shielding. Nevertheless, with minimal interference, well designed and matched heads are capable of very high quality reproduction.

Magnetic heads are of two basic types.

Moving coil

The stylus is coupled directly to the pick-up coil which is positioned between the poles of a permanent magnet. The stylus action causes the coil to cut across the magnetic field inducing a signal voltage. The coil size is a compromise between minimum mass and maximum flux linkage.

Variable reluctance—moving iron

The stylus is coupled to a ferrous armature which is allowed to move across a gap in a magnetic circuit. This varies the reluctance and therefore the flux in the magnetic circuit. Sensing coils positioned close to the gap have a voltage induced in them by this flux variation giving an output with a good response.

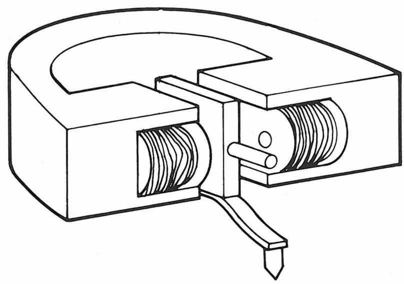
Moving magnet

The action is similar to that of the moving-iron type except that the moving element is magnetised not the pole pieces.

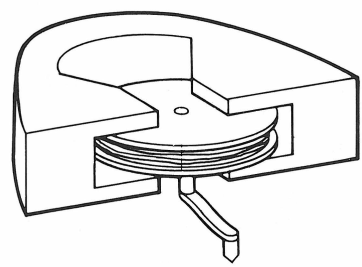
Induced magnet

The magnet and coil assemblies are fixed. A small ferrous armature, attached to the stylus, deflects the magnetic field inducing a current in the coils.

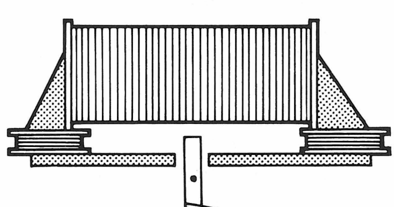
Basic outline of a moving-iron variable reluctance pick-up. The soft-iron plate pivots either at the centre or at the top. The sensing coils are inlayed in the pole faces.



Basic moving-coil pick-up. Typical values for this are: output impedance, 2 k to 5 k Ω at 1 kHz; and signal output, 2–5 mvolts at 1 kHz.



Variable reluctance pick-up using DC electromagnet and soft-iron needle-type armature. The illustration is diagrammatic and not actual. Typical values for this are: output impedance, around 10 k Ω at 1 kHz; and signal output, around 5 mvolts at 1 kHz.



Non-magnetic pick-ups have higher output impedances and signals than magnetic types.

Monophonic Pick-up Heads: Non-magnetic

The general mechanical–electrical transfer characteristic of non-magnetic heads tends to be amplitude rather than frequency conscious, the output being reasonably flat over a number of octaves. Output impedances tend to be high making stray capacitance a factor to be considered.

The crystal or ceramic pick-up

Basically the crystal or ceramic pick-up consists of a bimorphic crystal having piezo-electric characteristics, ie physical distortion of the crystal causes a voltage difference to appear between points on its surface. The crystal is sliced across its axes at carefully determined angles in order to obtain the wide-band transfer characteristic required. This determines the points at which the mechanical excitation must be applied and the output transducers attached to obtain the optimum performance.

The crystal is mounted in a ceramic container to protect it from moisture. The stylus is attached via compliant material to prevent mechanical shocks. The body of the crystal is immersed in a damping grease to reduce the effect of unwanted vibrations to which it is very susceptible.

These pick-ups have a very high output impedance with the result that stray capacitance and incorrect loading can impair the frequency response; the output signal level is a significant fraction of a volt. The crystal is relatively cheap to mass-produce.

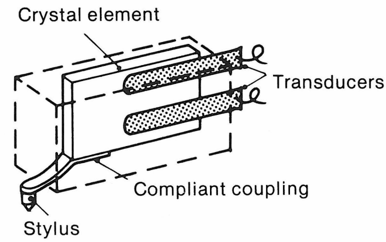
The semiconductor pick-up

Essentially a silicon-based transistor device to which the stylus is coupled with a compliant material, the semiconductor pick-up requires an external DC supply for its operation. Vibrations received from the stylus are applied in the region of a control barrier where it has a direct effect on the current flowing through the device producing an output signal voltage at high impedance. Correctly matched, this pick-up has an almost flat frequency response in excess of 30 kHz. It is available as a plug-in capsule.

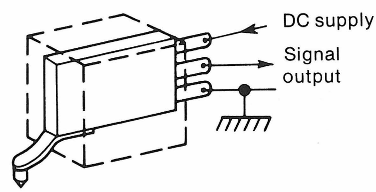
Light-operated devices

The development of fibre optics, lasers and the semiconductor photocell has revived interest in light-operated pick-ups for use in conjunction with 'hill and dale' or digital disc recordings. The pick-up makes no physical contact with the disc and so requires a sophisticated form of tracking control. One method uses a laser scanning system on static card recordings. Since there is no track wear such recordings remain unimpaired by use.

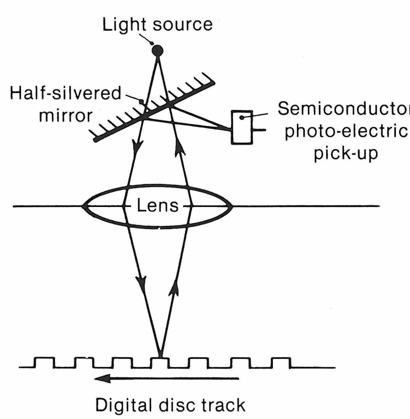
Crystal ceramic pick-up: Typical response 0–20 kHz; output impedance 1 M ; signal output 0.1–1 volt.



Semiconductor capsule. Typical response 0–30 kHz.



3, Light operated pick-up.



Stereophonic pick-ups have to resolve groove displacement in two directions at 90° to each other.

Stereophonic Pick-up Heads

The groove is tracked by a single stylus coupled to a resolving system so that the excitation from each wall is individually conveyed to a separate pick-up element.

In stereophony, the effect of crosstalk between the outputs is a reduction of the stereo effect. The track is recorded with a channel separation of 30 to 40 dB. It is inevitable that some crosstalk will occur in the pick-up head but a good quality head will give an output with a separation of the order of 20 dB.

Variable reluctance heads

The principle of operation is similar to that for monophonic variable reluctance heads (see p. 152) except that two sets of poles are used set at right angles to each other. Action of the armature along one axis activates only one circuit and vice-versa. Thus the outputs generated are more or less independent of each other. Heads of this type have been developed to a very high standard and are widely used.

Moving coil heads

The stylus is compliantly coupled to two small coils set at ±45° from the vertical in a single longitudinal magnetic field. The coil assembly pivots on a very precise centre axis; the armature couples at 90° to this axis such that the action of the armature at (say) +45° causes one coil to cut across the magnetic field generating an output whilst the other rotates around the lines of force with no effect. Thus each coil generates an independent output. For best results, the coil assembly needs to have small mass; the outputs, therefore, tend to be low level.

Crystal and ceramic heads

Crystal and ceramic heads consist, generally, of two crystal elements in a ceramic mounting at 90° to each other. They are coupled to the stylus via a parallelogram of compliant levers arranged so as to give the required flexing to each element. One bar of the lever system is fixed and the stylus is coupled to the opposite arm. The two elements are connected to the mid-points of the remaining two arms. The whole assembly is mounted in a ceramic case in a damping gel. Another type is in the form of a hollow cylinder with the stylus coupled to the internal surface and the transducers to the external.

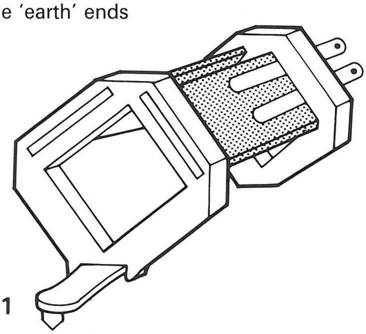
Semiconductor heads

Because of the wide bandwidths possible with semiconductor heads, they have proved to be eminently suitable for use with the single groove stereo-quadraphonic disc.

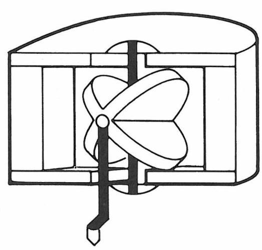
STANDARD COLOUR CODE FOR STEREO PICK-UP LEADS

No. of leads	Left Channel	Right Channel	Ground
3	White	Red	Black (common)
4	White, Blue*	Red, Green*	—
5	White, Blue*	Red, Green*	Black

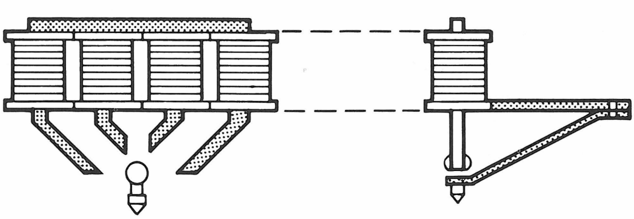
*Blue and Green are the 'earth' ends



1, Basic construction of a crystal stereophonic pick-up.



2, Basic moving coil stereo head showing the centre balanced cross-coil assembly with the armature coupled to the horizontal axis.



3, Outline of a variable reluctance stereo head. The side elevation shows the compliant arm supporting the magnetic armature.

The four signals required for quadraphony can be coded into two and recorded as a single groove.

Pick-ups for Quadraphonic Sound

The problem of reproducing the two separate channels that make up a stereophonic system has been solved by recording the two signals at right angles to each other and 45° to the vertical on a single groove.

For quadraphonic reproduction four discrete signals must be reproduced from a single groove while at the same time allowing complete compatibility to replay stereophonic records with the same equipment. An elegant solution to this difficulty was the introduction, in 1970 by the Victor Company of Japan, of the Compatible-Discrete Four system known as the CD-4.

The CD-4 system

Recording. The four sources are located at 90° intevals and nominated left (or right), front and back, ie LF, LB, RF and RB. These are combined to give four new signals, L(F+B), R(F+B), L(F−B) and R(F−B). The first two constitute standard stereophonic signals and are bandwidth limited to 15 kHz. The difference signals, L(F−B) and R(F−B), are separately modulated onto 30 kHz carriers and occupy the 20–45 kHz band. These are added to their respective 'low-band' summation signals to create augmented stereo signals which can be recorded on the opposite walls of a single groove.

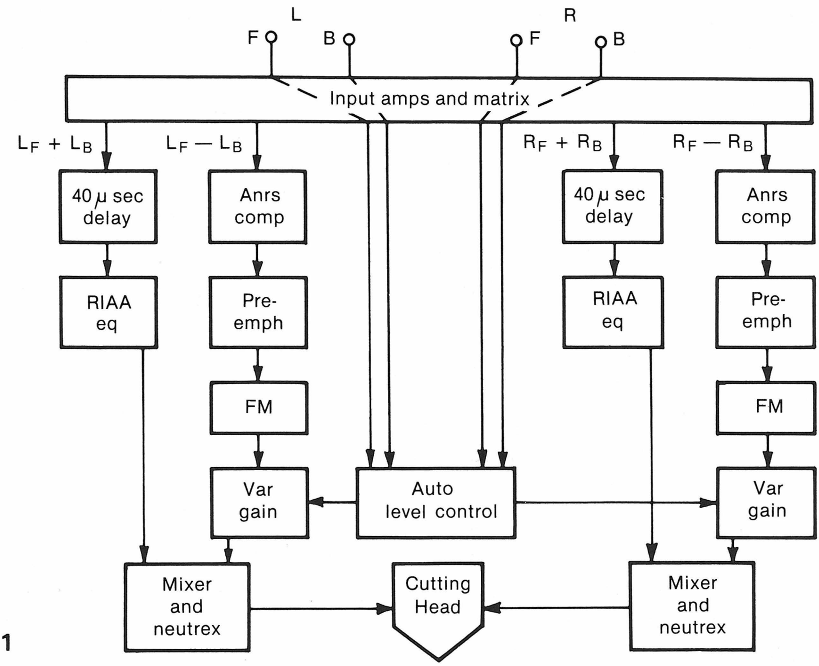
The difficulty of accommodating cutting heads to the wide bandwidth used is avoided by cutting the master disc at half speed.

Replay. Replay takes place, of course, at the correct speed and has necessitated the development of very high quality wide-band stereo replay heads in order to recover the complete signals with minimum crosstalk.

In order to minimise noise and crosstalk, the signals on the 30 kHz carriers are phase-modulated between 80 Hz and 6 kHz and frequency modulated outside this band.

Pick-ups for quadraphonic sound

The CD-4 and most other quadraphonic discs employ systems for multiplexing two channels into four. They are therefore compatible with ordinary stereo recordings and can be played using stereo pick-ups provided that they have a sufficiently extended high frequency response. The CD-4 system extends to 45 kHz and only the finest pick-ups, with very good compliance, can accommodate this range. Semiconductor heads have very wide bandwidth and are eminently suitable for the purpose.



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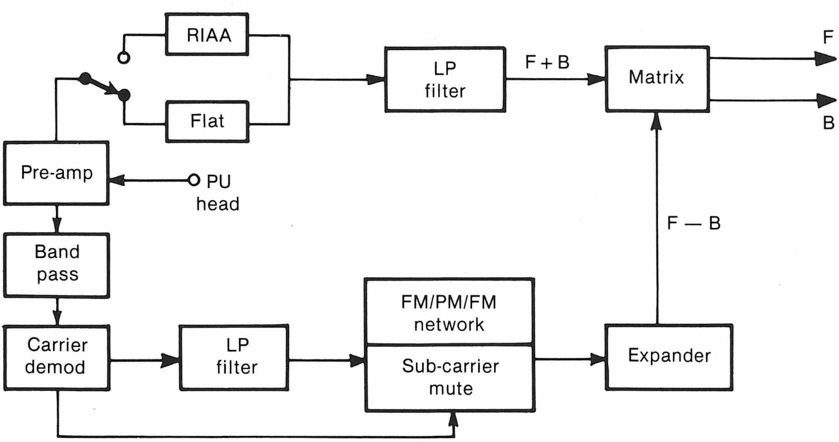

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1, Outline of signal processing for CD-4 recording; 2, Outline of one side of the demodulating circuit for the CD-4 system disc replay.

Pick-up Characteristics

During recording the groove deviation is restricted at low frequencies and boosted at high frequencies. This is done according to a standard laid down by the Record Industry Association of America and known as the RIAA characteristic. For a constant recording level this gives a constant groove deviation below 600 Hz and a groove (stylus) velocity proportional to frequency above 600 Hz.

This recording characteristic is used by most commercial disc manufacturers modified as being the combination of three separate curves defined by time constants, these being:

	Fine groove	*Coarse groove*
Treble:	75 μsec	50 μsec
Middle:	318 μsec	450 μsec
Bass:	3180 μsec	3180 μsec

An inherent advantage is the improvement in the signals:noise ratio since disc surface noise increases with frequency due to the structure of the material used.

Radius compensation

During recording high frequency boost is also used to compensate for stylus tracking errors due to the fact that the groove longitudinal speed decreases towards the centre of the disc. This is in addition to the RIAA characteristic.

Magnetic pick-ups

Magnetic pick-ups have an electrical output approximately proportional to the stylus velocity. Thus the pick-up output will closely resemble the recording signal after RIAA compensation and will need to be passed through a correction network with an inverse characteristic.

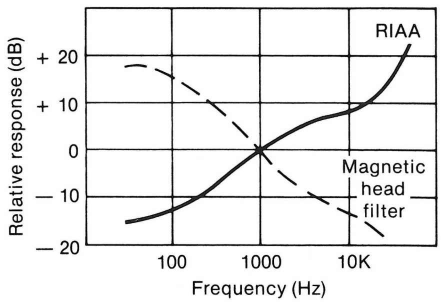
Crystal pick-ups

Crystal pick-ups, including semiconductor heads, have an output proportional to groove deviation. They will, therefore, have an output signal tolerably flat in response and do not require any correction if terminated in the correct impedance.

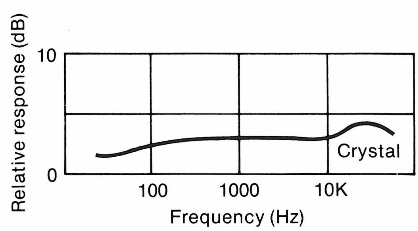
Connecting leads

High gains are generally involved in pick-up circuits, particularly in the case of high quality devices, and the connecting leads should be effectively screened. Attention should be paid to the capacitance of connecting leads as this may affect the high frequency response of the head. Some cartridge manufacturers assume lead capacitance in the head design; thus very low capacitance leads will affect their response.

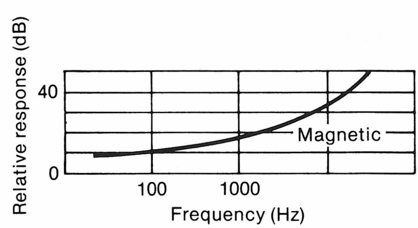
Relative response showing the RIAA recording characteristic and the inverse RIAA filter response required for magnetic heads.



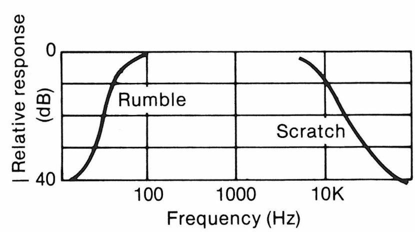
Typical response of a crystal pick-up.



Pick-up head output from RIAA Standard Test Disc.



Typical rumble and scratch filter responses.



Checking Gram Electrics

As with all electrical devices, disc playing equipment should be checked exhaustively for wiring and performance on installation. It should require only occasional checking thereafter unless the performance seems suspect.

Mains supply
Mains plugs must always be correctly wired according to the internationally agreed colour code: live = brown; neutral = blue; earth = yellow/green stripe. The earth lead must be correctly and securely connected at both ends. Fuses should be as prescribed by the manufacturer. Many turntables have two-pin mains connectors and a separate earth lead that goes direct to the amplifier.

Electrical performance
Special test discs can be obtained on which are recorded a series of tones covering the frequency band used in modern recording. Usually these are spot frequencies connected by a gliding tone; some include a special section at 1 kHz used for distortion and power output checks. They are manufactured very precisely to the RIAA specification and are designed for carrying out a complete check of pick-up and amplifier performance.

An audio frequency range measuring device fitted with high-impedance probes is required for use with the test disc.

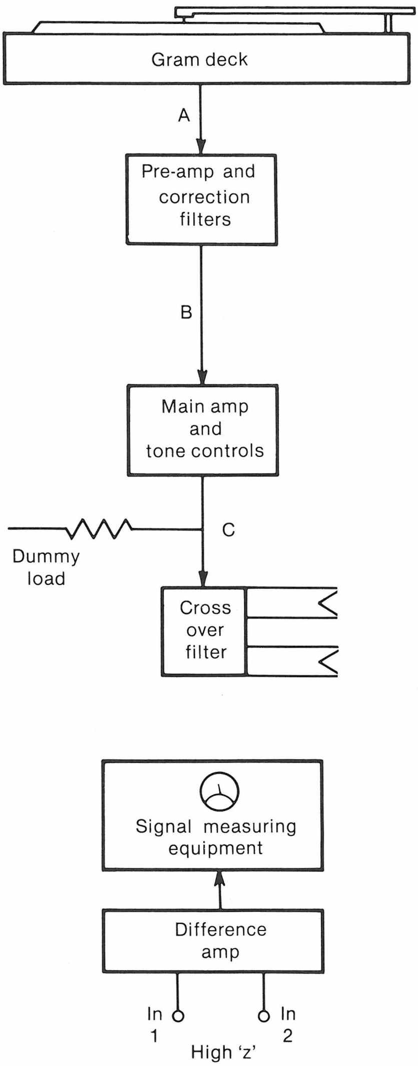
The pick-up. Using the test disc, the output of the head is measured at the input to any correction filters and checked against its specification. Measure the response again at the output of the correction filter and make any adjustments necessary to give the flattest possible response.

Pre- and main amplifier. Measure response at the output of the pre-amplifier with all tone and filter controls set for minimum effect. The effect of these controls on what should be a flat response should be measured independently.

The output of the main amplifier is best measured across a dummy load resistor representing the speaker system with the volume st at about three-quarters maximum. Distortion and power output are measured at maximum volume (or as specified) at 1 kHz.

The loudspeaker system. Where split speaker systems are used the characteristics of cross-over networks can be examined. The overall performance of the system is difficult to measure, however, any serious defects, such as cone-rattle or buzz, are easily discernible.

A record should be kept of performance checks for reference and comparison.



Performance measurements: *A*=pick-up response; *B*=pre-amplifier and filter circuit response; *C*=main amplifier response.

Turntable unbalance or incorrect tracking weight can cause unnecessary disc wear.

Checking Gram Mechanics

High quality disc playing decks are provied with a range of mechanical adjustments which should be checked when the equipment is installed and occasionally thereafter or when the location is changed.

The motor

The motor is mounted with rubber anti-shock mountings to prevent mechanical rumble being transmitted to the turntable. Although these do not normally wear, very occasionally they can break or perish which may cause the drive coupling to the turntable to go out of alignment. This is the usual cause of rumbling gears and worn or frayed drive belts. Some motors require occasional lubrication but self-lubricating bearings are quite common.

The turntable

The turntable is usually mounted on an escutcheon plate which is spring mounted onto the baseboard. With a spirit level on the turntable, the spring tensions should be adjusted until the turntable is level in all directions. Occasionally these springs can 'tire' causing misalignment. If the turntable is not level, it can cause wear of one side of the disc groove thus distorting the recording, particularely with stereo or quadraphonic discs.

Rim-drive turntables utilise a rubber pressure wheel which can develop flats if the pressure has been left applied when the deck is not in use. These flats cause an uneven drive speed which appears as wow. The only cure is replacement of the defective wheel.

The turntable revolves on a bearing race. If this is too dry or has accumulated dust and grit, it can cause rumble. The bearing should be inspected occasionally and cleaned and lubricated.

In all cases where lubrication may be necessary, the manufacturers will indicate when a preferred lubricant is advisable.

The tracking arm

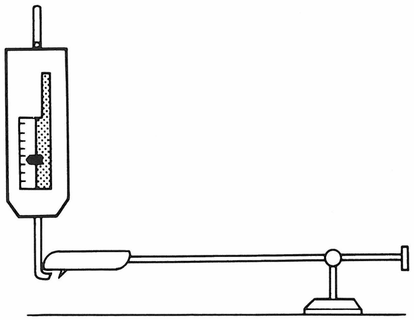
The pivot and balance should be kept clean and lubricated as necessary. Care should be taken to avoid getting any of the lubricant onto the electrical plugs and sockets for the pick-up.

Stylus pressure (tracking weight) should be adjusted using a spring tension balance to that recommended by the manufacturer. If plug-in heads are used, they should be checked each time the head is changed.

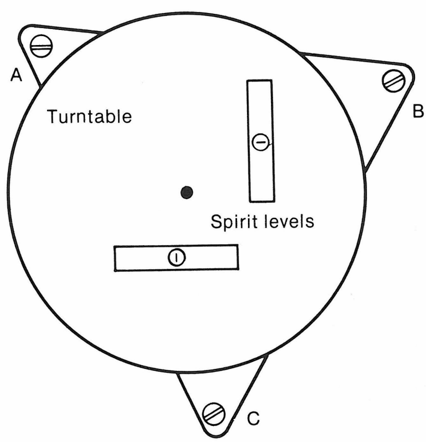
Incorrect stylus pressure can result in 'wall-riding' or skidding which will damage the disc as well as the stylus.

Stylus wear is best judged by examination under a microscope.

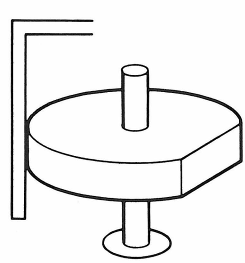
Checking tracking weight using a spring balance scaled in grams.



Levelling the turntable. *A, B* and *C* are spring tension adjustments.



Rubber rim-drive wheel showing a flat (exaggerated).



Convenience can be improved and wear on records and styli reduced by mechanical groove locating devices.

Groove Locating Devices

It is probably true to say that more gramophone records are spoiled and styli chipped by manually dropping the pick-up on the record than by any other cause.

Arm dropping mechanism

Even the simplest manually operated record player should be provided with a dashpot-damped arm dropping mechanism which, when a lever is operated, lowers the pick-up gently on to the required part of the record. This method of lowering the pick-up should *always* be used to preserve the record and stylus.

Groove selection

A somewhat specialised requirement, used in broadcasting studios, theatres and film dubbing, is the facility to locate a particular section of a record quickly and with some degree of accuracy.

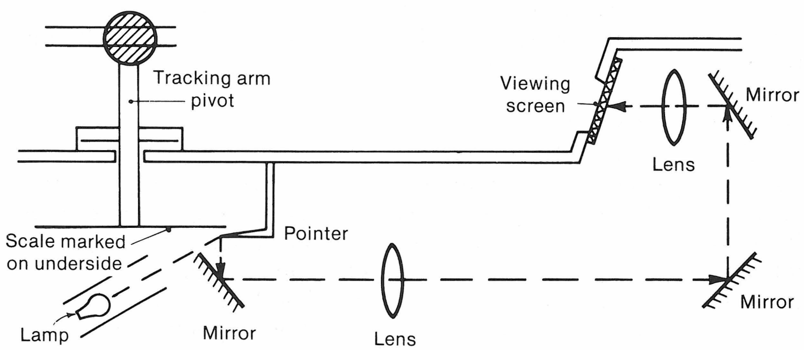
For this purpose groove locating devices have been made with which the position of the tracking arm is displayed on a ground-glass screen behind the turntable. Groove numbers can be read from the scale (a misnomer since there is, of course, only one groove) and related to the cues in the script or written on the record label so that the required position on the record can be located quickly. This position is, however, unlikely to be sufficiently accurate for many types of cue and it will be necessary to find the exact part of the groove (p. 168).

Automatic turntables

Although there is no longer the same need for autochangers as was the case with 78 rpm records, however, some degree of automation can be a considerable advantage—if only to reduce the risk of damage to the record through mishandling. Automatic turntables are available which can be set to play from a specific point or to sense the run-in groove position according to the size of the record. The majority of record players sense the run-out spiral (usually by the acceleration of the tracking arm), return the pick-up to rest and switch the motor off.

In the Accutrac 4000 system, the grooves are located by an infra-red source and detector in the pick-up head. It can be programmed to count the number of grooves as the head scans across them and lower the pick-up into the required position to play. It can be operated by remote control.

The trend towards linear tracking arms, some of which can be operated vertically, has made it necessary to automate the groove-locating process and has also made it easier to achieve.



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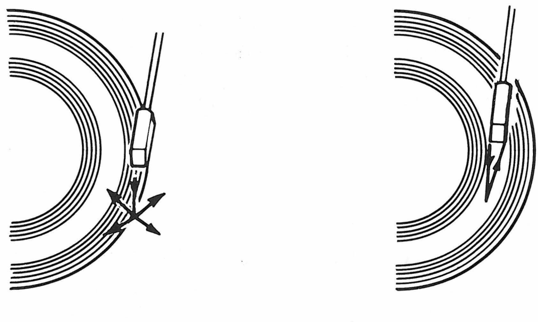

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1, Optical groove locating indicator. 2, Location of tracks by infra-red emitter and detector on pick-up. When Stylus passes over the grooves (a) the light is scattered, but when it focuses on the unrecorded 'land' it is reflected back into the receiver (b).

Professional work demands the ability to play records from predetermined cues.

Disc Cueing Methods

When records are used for professional purposes such as broadcasting, discotheque, theatre or for editing when dubbing records to tape, there is need for a high degree of accuracy in locating the starting point precisely on a time cue.

Locating the cue point

When an accurate starting point is required, it is unlikely to be found by dropping the pick-up into the groove (see p. 166) due to the spiral nature of the groove not to mention the possible eccentricity of the spindle hole. The procedure is, therefore, to lower the pickup into the groove ahead of the cue (listening on 'prefade'). When the cue is heard the turntable is stopped and rotated slowly backwards by whatever angle of rotation it requires to get fully up to speed.

If this is a frequent requirement, especially where speed of action is involved as in the professional field, the pick-up cartridge should be chosen for its ability to stand up to backwards rotation.

Quick-start turntables

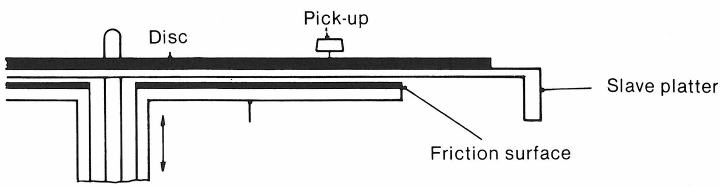
To achieve accurate cueing by the above method it is necessary to have a turntable which comes up to speed very quickly, otherwise every start will be marred by 'wow', especially in the case of music. There are two basic methods of obtaining a quick start: mechanical and electronic. The mechanical system consists essentially of two turntables arranged one above the other. The lower turntable, which has a heavy flywheel action, rotates continuously and can be raised or lowered by means of a lever or a solenoid actuated by a contact on the backstop of the gram fader. The upper turntable, the platter, holds the record; it is undriven and at a fixed height.

When the cue is required the channel is faded up, the solenoid raises the turntable to contact the platter which takes up speed very quickly. Alternatively, some quartz-locked servo-operated turntables (p. 144) have high-torque motors and provide very rapid start and speed stability (well within $\frac{1}{4}$ revolution) and are eminently suitable for cueing purposes.

The slip mat

A cheap method for quick-starting is the slip mat. This is a piece of material with a suitable friction coefficient which is placed between the platter and the record. The turntable is set in motion and the mat held still with the fingers holding the disc until the cue.

1



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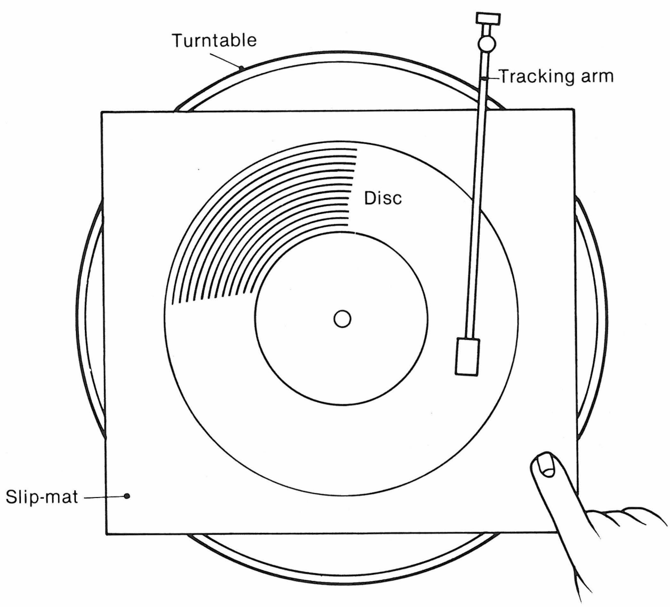


1, Quick-start professional turntable. The turntable rotates continuously and is raised to friction-drive the record platter. 2, The slip mat fast start technique.

Microgroove discs are made of soft material and must be treated with care.

Care of Discs

The modern microgroove disc is usually a vinyl pressing from a master negative. Vinyl has the requisite physical properties for obtaining prints of excellent quality. Unfortunately, these very properties are a source of problems in the life and use of such discs. The material has a soft surface which has the tendency to accumulate static electrical charges which attract dust particles; these are liable to lodge in the very fine grooves and can cause the stylus to jump and mistrack. The surface is also very susceptible to scratching which can ruin the quality of the recording. Indeed, the surface of the disc should not even be touched by the bare fingers, since the grease deposited can clog the groove and help to cement-in particles of fine grit. Fine cotton gloves should be worn if many discs are to be handled.

Microgroove discs should never be allowed to come into contact with each other or any rough surface. They should not be stacked for playing on a 'record-changer' machine.

Storage

The disc should only be removed from its protective covering when it is to be used; it should be replaced immediately after use.

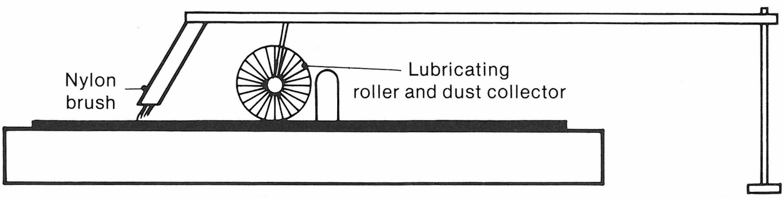
Discs should not be stacked in piles as the collective weight can distort the grooves of the discs at the bottom of the pile. The best way to store discs is inside their protective covers, in individual vertical slots of a filing unit away from sources of heat and damp; they should not lean against each other.

Cleaning and anti-static treatment

Cleaning should be carried out with a soft anti-static cloth. There are several types of the 'dust-bug' variety which clean and lubricate the disc whilst it is on the turntable. Such devices should not be attached to tracking arms which have not been designed to accommodate them; the electrical tracking performance of the head may be affected by the necessary tracking weight of 5–10 g. Dust-bugs incorporate some form of soft nylon brush composed of fine fibres mounted on an independent arm; another brush or roller is also attached to collect the dust and at the same time apply a controlled amount of anti-static lubricant. A static discharge devise is available in the form of an aerosol spray.

Lubricants are made from soaps, alcohols and fatty acids in a volatile solvent which will evaporate leaving the lubricant as a very thin skin. Its presence helps to reduce the surface noise of the disc.

Although groove wear is inevitable, careful handling, cleaning and lubrication can increase the life span of the disc and styli by an appreciable factor.



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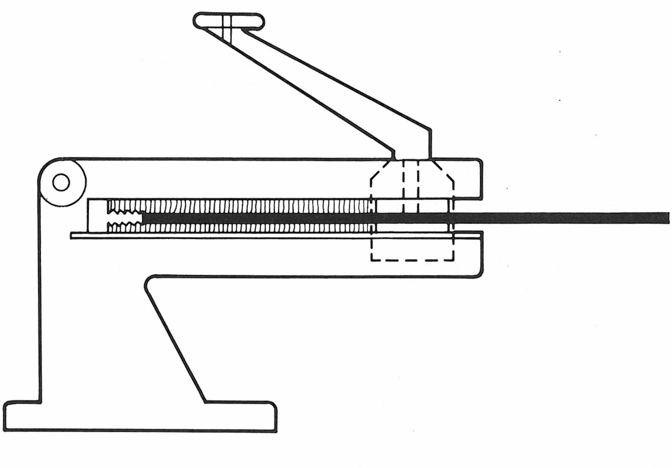


2

1, A 'dust-bug' device mounted on its own arm can be used while the disc is being played; 2, Example of a commercially used disc cleaner and lubricator. There is a handle for rotating the disc; other versions have a small electric motor to drive the disc. The disc is clamped lightly between cleaning and lubricating pads.

Digital techniques offer considerable improvement in audio quality over conventional analogue systems.

Digital Sound

Sound recording has advanced to a high level of excellence using analogue techniques, ie systems in which the signal strength is related to the original sound volume. This is still not perfect and the law of diminishing returns dictates that further advance would be very costly. The future trend is, therefore, towards the use of digital techniques.

Digital techniques

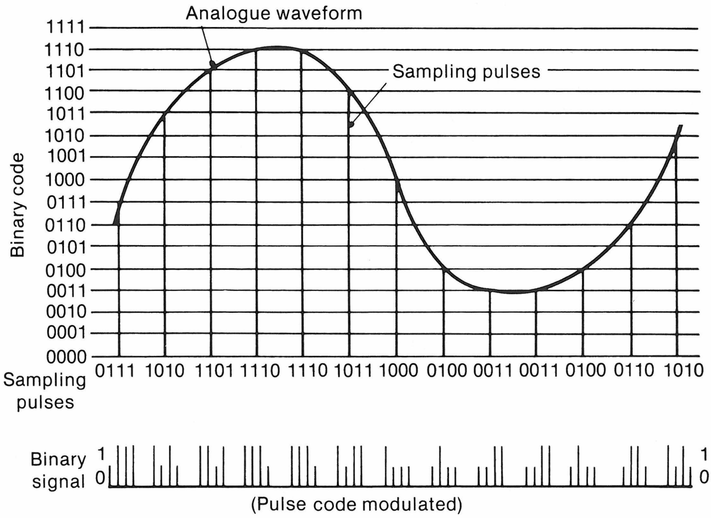
In a digital system the original, analogue signal is converted to a code composed of a series of pulses. The pulses represent binary numbers corresponding to the amplitude of the waveform at each moment in time.

Binary numbers are composed of only two values, 'on' (logic 1) or 'off' (logic 0). The pulses can be at either full amplitude or not at all.

Analogue to digital conversion

The process of converting from analogue to digital format consists of:

1. *Sampling*. Samples are taken of the instantaneous value of the analogue waveform. They are taken at a fixed rate which must be at least twice the highest frequency that it is intended to convert. Each sample is held in a store while it is measured and its value converted into a code.
2. *Quantising*. The process of measuring each sample involves comparing it against a scale consisting of a number of discrete values called 'quantising levels', each one of which is assigned a reference number to represent its value. As the analogue signal can have values which vary continuously over the whole range of measurement, the actual level and the quantising level will seldom coincide exactly. The greater the number of quantising levels the more accurate will be the measurement. Differences between the sample amplitude levels and the quantising levels can be heard as noise in the reproduced sound. As the error is proportionately greater with fewer quantising levels, the noise tends to be more objectionable at low levels.
3. *Coding*. The quantising levels are converted into a code, usually binary, which represents them as a group of digits composed of 'on' or 'off' signals. The number of quantising levels required for high quality reproduction would normally be in excess of 1000. A binary code composed of 10 digits (bits) will provide 1024 (ie 2^{10}) quantising levels with a peak signal:peak quantising noise ratio of 60 dB. A 13 bit code would allow 8192 levels with a signal:noise ratio of 78 dB.



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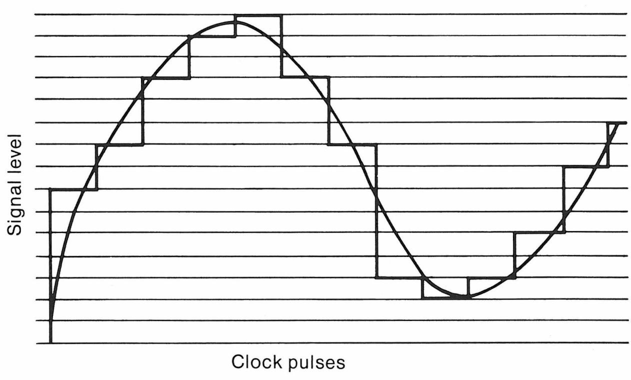


2

1, An example of sampling and coding of an analogue signal. For simplicity only a 4 bit code is shown which would allow only 16 levels of measurement (quantising levels). This would result in a high degree of error and thereby a high level of quantising noise. This takes the form of 'white noise' (ie noise with an equal spectral distribution) on large amplitude signals and 'granular distortion' (resembling non-linear distortion) at low levels where only a few quantising levels are in use. For high quality sound reproduction, codes with about 13 bits are used. A 13 bit code would provide 8192 quantising levels and a signal:noise ratio of 78dB. Some systems use logarithmic instead of linear sampling so that there are more quantising steps at low level where it is more critical; 2, Digital to analogue conversion. The digitised signal is converted to a series of levels corresponding to the original samples presented at intervals ('clocked') corresponding with the pulse rate of the original sampling process. This produces a jagged waveform which, after passing through the low-pass filter, resembles the original analogue wave.

Digital disc recordings can offer immaculate frequency response with very low noise.

Recording Digital Sound

On the previous page the method of converting a normal, analogue signal to a digital code was described. The process involves sampling the signal at a rate of at least twice the highest frequency required to be converted and describing these levels as a series of digits (called bits). The higher the number of bits the better the signal: noise ratio.

Bandwidth
Unfortunately the high quality and freedom from noise of a digital system can only be achieved at the expense of bandwidth. For example an audio bandwidth of 15 kHz and a signal:noise ratio of 78 dB (13 bit code), assuming a sampling frequency of 2.2 times, requires a bit rate of $13 \times 15\ 000 \times 2.2 = 429\ 000$ which suggests a bandwidth of 429 kHz. Such a frequency range puts it outside the range of normal audio recorders but well within the scope of video and data recording systems.

Video cassette/sound recorders
In the case of the video cassette recorder where the consumer has the opportunity to make recordings as well as reproduce them, there is a strong argument for the compatible machine that affords the choice of video or very high quality sound. Convertors are available that can convert a standard VCR for the purpose.

Digital disc recording
The most convenient playback medium is the disc. Disc systems have been evolved for replaying prerecorded television programmes using digital technique, with a bandwidth of the order of 4 mHz. Such a system is clearly adaptable either as a digital audio player or as a video player with audio capability.

Disc size
In the absence of a universally accepted standard for disc recording the various systems are incompatible with each other. Disc sizes include 11 cm compact disc and the 30 cm discs which are similar to those used for video. Playing times (for stereo) vary between 30 and 150 minutes.

Quadraphony
Some systems offer the facility to code four signals (for quadraphony) by halving the playing time.

COMPARATIVE FEATURES OF SOME DIGITAL AUDIO DISC FORMATS*

Make	*Purpose*	*Signal system*	*Tracking*	*Speed*	*Disc diameter*	*Playing time*
Sony	Video/Audio	Piezo	Electronic (non groove)	450 rpm (audio) 900 rpm (video)	12 in (30 cm)	$1\times2\frac{1}{2}$ hours
Philips VLP	Video/Audio	Optical	Electronic	1500–600 rpm Linear writing speed.	12 in	2×1 hours
Philips compact disc	Audio only	Optical	Electronic	215–500 rpm Linear writing speed	$4\frac{1}{2}$ in	Single sided 1×1 hours
JVC VHD/AHD	Video/Audio	Capacitance	Electronic	900 rpm	12 in	2×1 hours
Matsushita Visc-11	Video/Audio	Piezo	Mechanical (groove)	450 rpm	12 in	2×1 hours
Matsushita Visc-O-Pac	Video/Audio	Capacitance	Mechanical	300–700 rpm Linear writing speed.	9 in 7 in	$2\times1\frac{1}{4}$ hours $2\times\frac{1}{2}$ hours
RCA Selectavision	Video/Audio	Capacitance	Mechanical	450 rpm	12 in	2×1 hours

*While it is the policy of this book to avoid mention of specific equipment, because of the present unsettled state of the art, the accompanying table is included with details of some of the current digital audio disc systems. Clearly the digital audio (or video) disc will not become commercially viable until standardisation can be agreed.

Digital recording can be stored on disc in the form of shallow pits which can be read with a laser beam.

Optical Digital Disc Reproducers

For digital recording the original (analogue) signal is converted to a code which represents the instantaneous values of the signal at a series of moments in time. The code is composed of binary numbers which can be represented by a series of 'on' (logic 1) or 'off' (logic 0) signals.

The laser-read disc

One method of representing a binary code signal is in the form of tiny pits in the surface of a disc. The presence of a pit can represent logic 1 and its absence, logic 0. The pits are arranged in a spiral formation so that they can be tracked like a gramophone record except that electronic tracking is used instead of a stylus. After formation the pits are covered all over with a transparent plastic to protect them from dust and handling distortion.

A laser beam is used to read the pits. It is focused on them through the transparent plastic carrier from underneath the disc. The pits are coated with a reflective surface which reflects light from the laser back along its optical path where it is deflected by a half-silvered prism into a photo diode. When there is no pit the light is defracted beyond the angle of the lens so that the light returning along the optical path is greatly reduced. The output of the diode is fed to a digital/analogue convertor to provide the analogue output. The return laser beam is also fed to two spots on the photodiode to provide tracking and focus error signals.

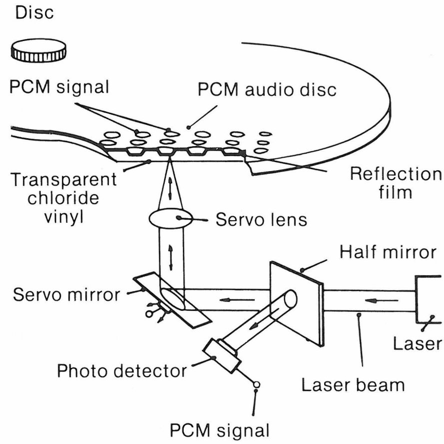
Tracking

Recording a PCM signal requires the storage of an enormous amount of imformation, albeit in the simple on/off form. In fact, an audio programme one hour long requires information of the order of 6 billion bits. To achieve this on a single disc, the packing density has to be high.

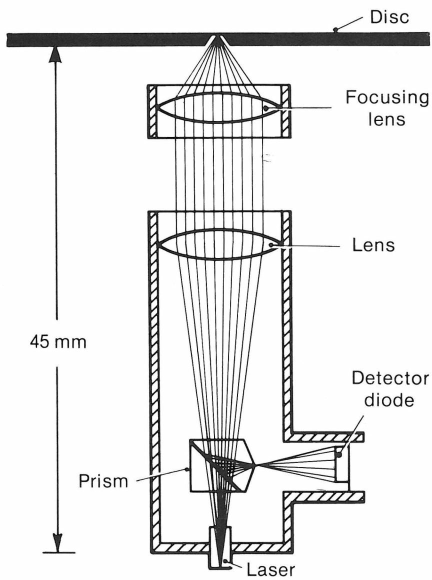
In the Philips compact disc system the 'grooves', ie the separation between the helical train of pits, is only 1.66 μm and to achieve maximum packing density the tangential velocity of the disc and therefore the spacing between the bits is maintained constant. This means that the rotational speed of the record has to be continuously varied as the laser tracks from the inside to the outside of the disc.

Some other systems employ constant rotational speed at the expense of length of recording/size of disc; the speed is just fast enough in the centre and faster than necessary on the outside.

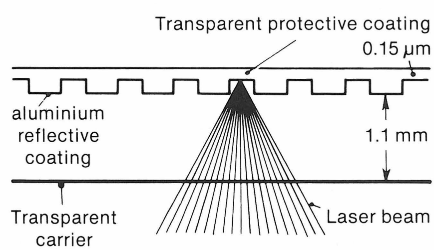
Representation of the Philips VLP system. The inset shows the pattern of pits in the record (enormously enlarged). The pits appear white. The LP record is played from underneath. The spring-suspended lens automatically focuses the laser light beam. A hinged mirror is used for following the track.



Optical pick-up of the Philips compact disc.
The laser light is focused on the aluminised reflective coating at the depth of the pit and is reflected back along its path to be diverted by the prism into the detector. Where there is no pit the beam is defocused and much less light is returned.



A greatly enlarged section through the disc. The 1.1 mm transparent carrier is the part that is pressed like an ordinary gramophone record and coated with reflective aluminium. The 0.15 µm transparent protective coating is applied later.



Audio PCM disc players have been designed using very fine groove mechanical systems.

Mechanical and Capacitative Digital Discs

Apart from the optical recordings described on the previous page, there are various other prototype disc systems designed to replay video signals which can be adapted for PCM audio reproduction.

The Teldec video disc

Among the first practical video discs was a mechanical design developed by AEG-Telefunken and Decca. This is similar to an ordinary gramophone record except that there is no 'land' between the grooves which are recorded vertically. The groove width is only 2.5 μm so there are about 50 times as many grooves per millimetre as in an average stereo LP. The signal is frequency modulated with an upper frequency of 3.75 mHz. This is too high for a normal stylus to follow so a system of 'pressure scanning' is used in which a skid-shaped scanner rides on a number of undulations simultaneously. These are compressed against a cushion of air on which the disc rests and the sudden jerk when they are released at the back of the stylus produces the movement which is translated into a signal by the ceramic transducer.

The Matsushita electric mechanical disc player

The Visc-11 format developed by Matsushita (technics) is a mechanical system similar to the Teldec disc but with finer grooves played with a diamond stylus. The track pitch is 2–3 μm giving one hour playing time on a 12 in record.

One problem with using a very fine groove system is the risk of handling damage and dirt. To counter this Matushita have produced the Vis-O-Pac, a 7 or 9 in disc which is maintained within a plastic cover from which it is removed only in the player. The Vis-O-Pac uses a constant linear writing speed (see previous page) working between 300 and 700 rpm to give $1\frac{1}{4}$ hours of audio playback on the 9 in and $\frac{1}{2}$ hour on the 7 in record.

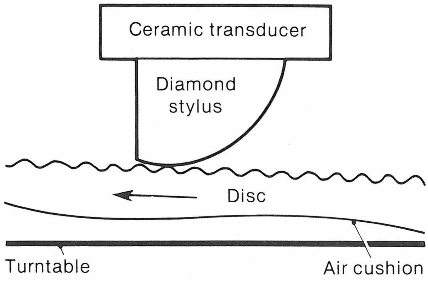
The capacitive disc

The Victor VHD/AHD is a video reproducer which can also be used for digital audio with an external PCM adaptor.

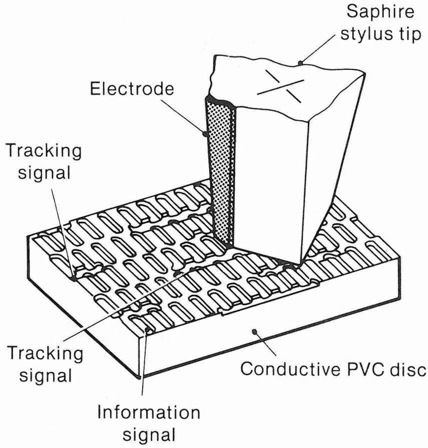
The disc has no grooves and operates on the capacitance principle. Two tracks are recorded, side by side, in the form of shallow pits in a conductive PVC. One track carries the audio/video information and the other provides a tracking signal which enables the sapphire stylus which has a blunt tip to follow the track. The signal is generated by the change in capacitance between an electrode on the stylus and the record. The tracking signal makes possible a random access selection for particular groove locating.

Selectavision (developed by RCA) is similar to the Victor VHD/AHD but has mechanical tracking (stylus and groove).

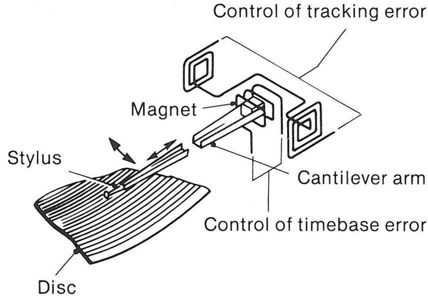
Greatly enlarged view of the stylus of a mechanical video disc system. The stylus is in the form of a skid which compresses several undulations simultaneously, distorting the record which is supported on a cushion of air. The action of the undulations 'popping up' behind the stylus causes it to vibrate.



The surface of a JVC capacitative disc. A tracking signal, recorded alongside the information signal, guides the blunt-tipped sapphire stylus which rides on the disc. The variation in capacitance between the conductive electrode on the stylus and the conductive PVC disc constitutes the output.



The JVC stylus is mounted on a cantilevered pick-up arm with a small magnet at the other end. A single coil surrounds, but does not touch, the magnet and a pair of vertical coils (wound in phase opposition to each other) are mounted on either side. Control voltages derived from the tracking error signals on the disc enable the stylus to follow the information track precisely. It can also be programmed to move the stylus to a specific track position.



Video recorders are used to record audio signals as a complement to vision or, using pulse code modulation, as very high quality sound recorders.

Sound on a Videotape Recorder

The main problem concerned with recording video signals is the very large bandwidth required. This can extend from DC to over 5 MHz. It has been solved in the case of videotape recording by two artifices: frequency modulating the signal and increasing the recording/reproducing head-to-tape speed by mounting the head (or heads) in a drum which rotates obliquely in relation to the movement of the tape.

Transverse recording

The established format for broadcast and, until recently, the only system with sufficient time base stability, has been the quadraplex transverse recording format. This uses four heads rotating virtually at right angles to the movement of the 2 in wide tape. The heads are switched as they rotate to lay down the video track across the tape. Approximately 16 consecutive tracks make up one field of picture. The audio signal cue track and control track are arranged longitudinally along the length of the tape. The magnetic particles in the tape are orientated across the tape to promote video quality and this, coupled with the comparatively narrow track width, tends to be a limitation on the quality of television sound.

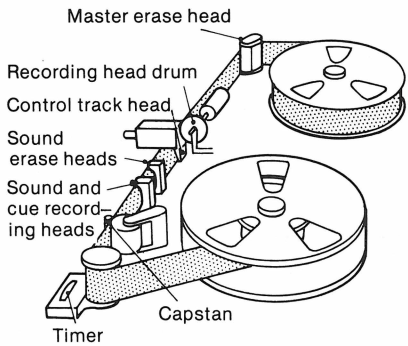
Helical scan recording

Helical scan recorders, in which the tracks are recorded diagonally with one field to each track, were originally designed for the domestic market but, with the advent of digital time base correctors (which can restore the time base stability), are finding increasing application for professional purposes.

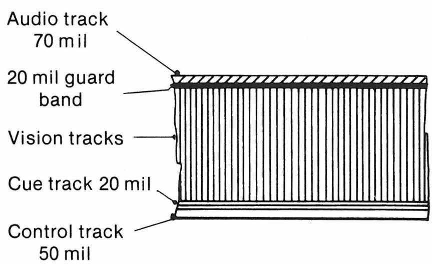
Unfortunately, unlike quadraplex format recording, where there is a universally accepted standard, there are a number of different types of helical scan. Some use one head, some have two, some record on 25.4 mm wide tape, others on 12.7 mm. There are also several different tape wrap configurations. Single head machines, where the head records a complete field on each rotation of the head drum, are capable of still-frame or slow-motion playback. Two head, segmented scan machines can be used for this purpose in association with a field store.

Audio tracks

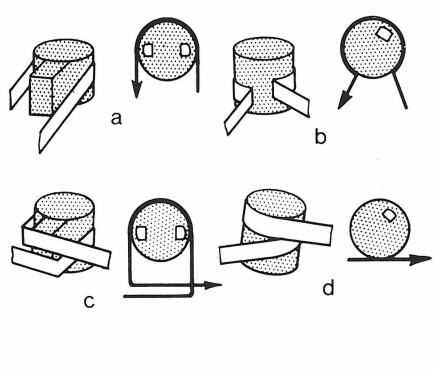
The audio recording tracks, control track and cue and address tracks are arranged longitudinally along each edge of the tape. One format incorporates five audio frequency tracks, two programme and one cue track on one side of the vision tracks, with the control and address-code tracks (for editing control) on the other.



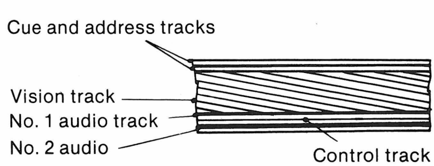
General arrangement of quadraplex videotape recorder.



Quadraplex recording on 2 in tape. The control track acts like a series of 'electronic sprocket holes' to synchronise the movement of the tape and the headwheel.



Various headwheel configurations. *a*: two-head alpha wrap; *b*: single-head omega wrap; *c*: two-head alpha wrap; *d*: single head alpha wrap.



Helical scan recording.
One helical scan format incorporates five AF tracks—two programmes and one cue track on one side of the vision tracks with the control and address-code tracks on the other.

Digital recording gives exceptional quality and freedom from noise.

Professional Digital Audio Recording

Professional analogue sound recording has reached a very high degree of excellence but there remain three important limitations: *a*, achieving a sufficiently high signal:noise ratio, especially where a number of sequential copies have to be made; *b*, eliminating modulation noise, lack of clairty in the extreme high frequencies which appears to put a veil over the sound; and, *c*, obtaining a clean extended bass response. Choosing a high tape speed, ie 30 rpm, to improve HF response creates problems in achieving bass because of the long wavelength: head gap ratio.

Digital tape recording

Digital recording on tape can offer a frequency response, flat within half a decibel from DC to 20 kHz (in practice the bass is rolled-off to 20 Hz) with a signal-to-noise ratio of around 90 dB and complete freedom from wow, flutter and modulation distortion.

The tape recorder can be a specially designed audio recorder (if multi-tracking is required) or a standard video recorder with a PCM adaptor.

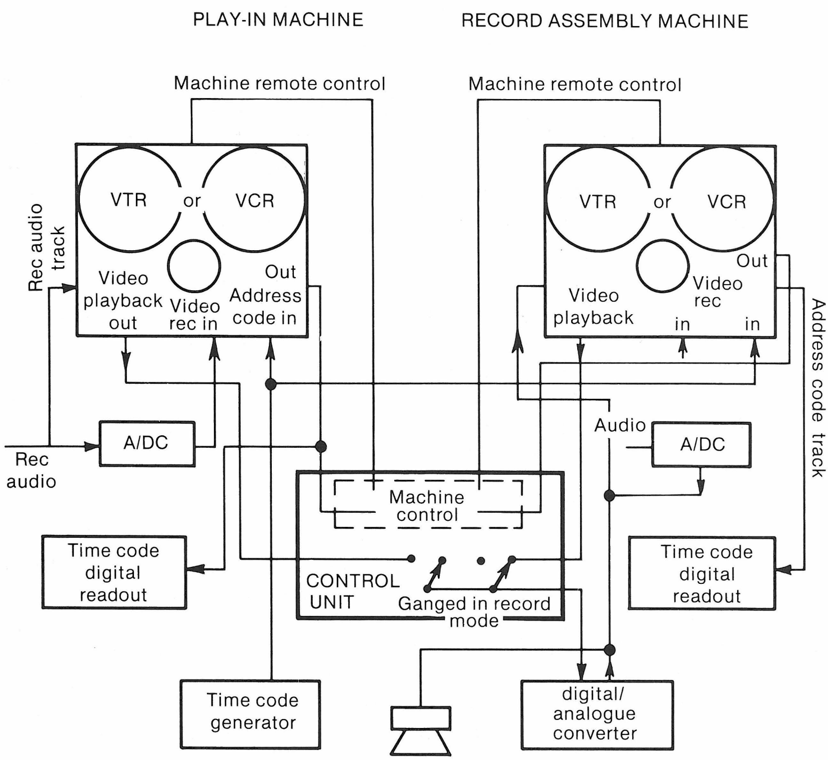
Editing

Editing a digitally recorded tape is not a simple matter of cutting and splicing it as with analogue tapes. Digital technique has much in common with video recording in that both require a stream of high density, high rate information which must be related to a continuous time base (synchronising or clock pulses). It is not possible to cut and join two video recordings except between picture fields and then only if the proper relationship exists between the two fields to be joined. Cutting and joining helical scan recordings is impracticable because of the long slanting nature of the tracks.

Video and digital recordings can be edited by dubbing from one machine to another. To do this requires a sophisticated electronic synchronising system which involves recording a digital time code on the cue or address track of both machines. This acts like sprocket holes with film except that every one has a discrete number. The machines can then be run 'locked' together at the edit point maintaining the proper sinc pulse relationship throughout. The new sequence can be butt-joined or cross-faded to the previously recorded one (normal diagonal cut editing is a form of cross fade) and levels adjusted in the process. As the recording medium is digital it suffers no degradation in the dubbing process.

Cue and address code

Using a standard VCR the audio signal can also be copied to the normal (longitudinal) audio track to help in identifying the sequences; the address code, on another longitudinal track, can provide a digital display of tape position.



Schematic representation of two-machine time-code editing for digital sound. The control unit controls the mechanical functions of the playback machine and the record machine. A time code recorded on the address code track of each tape gives a visual indication of their position. When both tape machines are locked to the synchroniser in the control unit they can be run in replay, record or spool in perfect sync. The synchroniser can also be used to locate a particular position on the tape. Before editing, the machines automatically set to a position a few seconds ahead of the editing point. Editing is achieved by running the two machines together and playing in the selected sequence at exactly the required edit point on the master tape set by the time code. The edit can be checked by running through the transition in the 'rehearse' mode, when only the monitoring changes over at the edit point. If this is not exactly right, the edit point can be moved (by offsetting the electronic lock-up) and only when it is considered perfect is the device put into the 'record' mode. When the edit point is reached, the record machine goes into record and the edit is completed.

Multi-track recording is used to save artists' recording time and for synchronising purposes.

Multi-track Recorders

Professional master tapes are nowadays nearly always made on multi-track recorders. These may be 4 or 8 track machines using 25.4 mm (1 in) tape, or 16, 24 or 36 track on 50.8 mm (2 in) tape.

The purpose of multi-tracking

There are three basic reasons for multi-tracking:

1. To enable re-mixing and various forms of manipulation to be accomplished at a later date without wasting recording time.
2. To enable artists to play in accompaniment to their own performance, to increase the apparent number of artists on the final recording, allowing for repeat takes of each addition without spoiling the previous ones and without unduly increasing the noise level.
3. To enable recording machines to be synchronised with other tape recorders, videotape recorders or film machines, using one track to carry a synchronising signal or time code. In this way sound tracks for video recordings or films can be built up, adding dialogue, effects, music etc. on the various tracks all synchronised with the picture. They can then eventually be mixed down to form a single synchronised sound track.

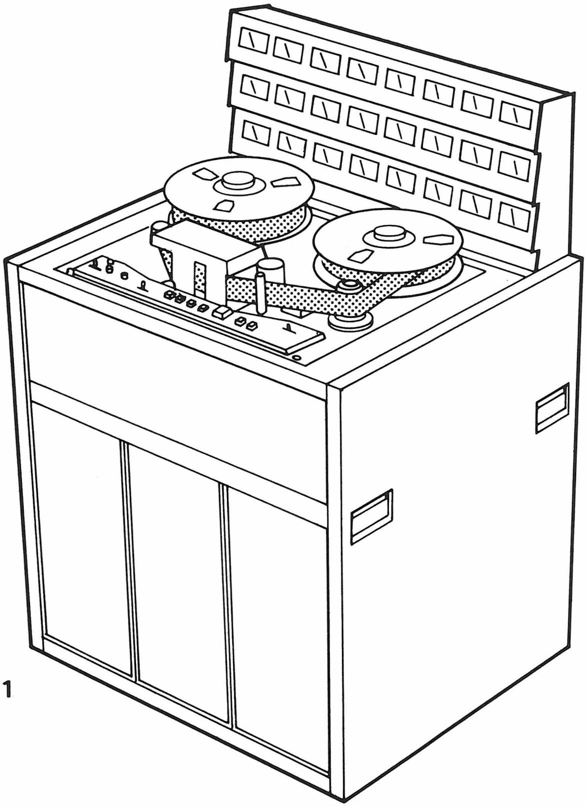
Multi-track recording

Making a multi-track master recording involves multi-microphone technique. Each section of the orchestra or instrumentalist in a group is given a separate microphone, sometimes more than one to each player (eg a drum kit can require up to 10 microphones for some types of balance). The microphone outputs are processed in a multi-channel sound console which provides the facility to monitor individual channels (by selection) or produce a 'dummy' mix in which they are mixed into the monitoring system while remaining separated for the recording tracks. Visual monitoring is provided for all channels so that the volume on each of the tracks can be kept within the parameters of the recording system.

At a later stage the various tracks are 'mixed down' to mono, stereo or quadraphonic form. This process can take a considerable time if much editing is involved or complicated treatment (eg various forms of artificial reverberation and frequency response shaping) is required for the individual tracks. It can all be accomplished, however, without retaining the artists.

Multi-track recorder mechanisms

Tape recorders handling wide tapes require powerful motors and well engineered mechanical systems and electronic tape tension sensors because of the large transfer of weight as the tape spools from one reel to the other.



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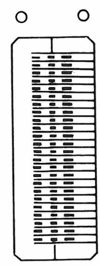


2

1, A typical 24 channel multitrack recorder. Note the VU meters for each channel.
In order that the tracks can be recorded at different times but in sync it is necessary to employ a complicated system of head switching. Individual erase heads are provided for each track.
Although separate record and replay heads are provided, arrangements have to be made to use the record head to playback previously recorded tracks while further tracks are being recorded to enable the artist to synchronise with the previous recording.
2, Typical multitrack record head. Great care must be taken in setting the azimuth otherwise serious phase differences can exist between the most separated tracks.
When selecting tracks for recording it is advisable to place any sources that are likely to have overlapping sound (eg two microphones in close proximity) on adjacent tracks. Otherwise phase differences could cause undesirable effects.

The last, and often the weakest, link in the sound reproduction chain is the loudspeaker. Problems stem from the wide range of wavelengths involved.

Loudspeakers

Moving coil loudspeakers

The vast majority of present day loudspeakers are of the moving coil type. These consist essentially of a diaphragm attached to a small circular (speech) coil which is suspended in a small annular gap between the polepieces of a powerful magnet.

Alternating current applied to the coil produces a varying magnetic flux around it. This flux interacts with the field from the magnet to cause it and the attached diaphragm to move in and out.

The diaphragm can take the form of a dome within the circumference of the coil and/or a cone surrounding it. The dome shape acts as a piston and is suitable for high frequency reproduction. The action of the cone is more complex and varies with frequency. The cone only behaves as a true piston for frequencies with a wavelength well below half its diameter (typically about 400–500 Hz). As the frequency increases the radiating area moves towards the centre, the outer area of the cone acting as a flexible mounting and as a horn-loaded termination. The contour of the cone and the stiffness of its construction, therefore, has a considerable bearing upon the upper frequency response. The low frequency response is largely conditioned by the mass/stiffness of the cone, the flexibility of the outer and inner suspensions and, of course, the baffle in which it is mounted.

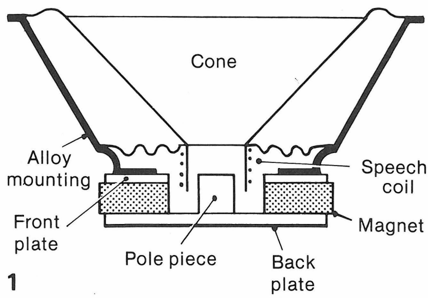
Multi-unit loudspeakers

The problem of maintaining loudspeaker diaphragm efficiency throughout the audio frequency range has led to the introduction of multi-unit loudspeakers in which the spectrum is divided into two, three or four frequency bands each handled by a different unit specifically designed for its range. Large cone loudspeakers with free suspensions allowing large cone movement are used for the bass, smaller cone units are used for the mid range, and small flat or dome-shaped diaphragm units for the high frequency. The loudspeaker units are each supplied via a crossover filter network.

Motional feedback loudspeakers

Reasons of economy dictate that most loudspeakers have passive crossovers, ie after the power amplifier, but this creates problems of matching. Loudspeaker impedances vary, increasing with frequency. An elegant way of overcoming this problem is to use separate amplifiers for the units in which the feedback circuits of the amplifier are controlled by the voltage developed in a special caramic pick-up mounted on the cone but not physically connected to anything else. The control emf is produced by the rate of acceleration of the cone acting against the inertia of the pick-up. The output thus compensates for loudspeaker resonances and environmental effects.

Cross section of typical moving coil loudspeaker.



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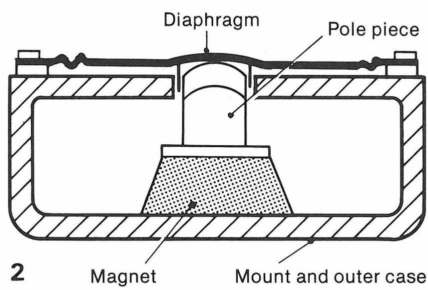
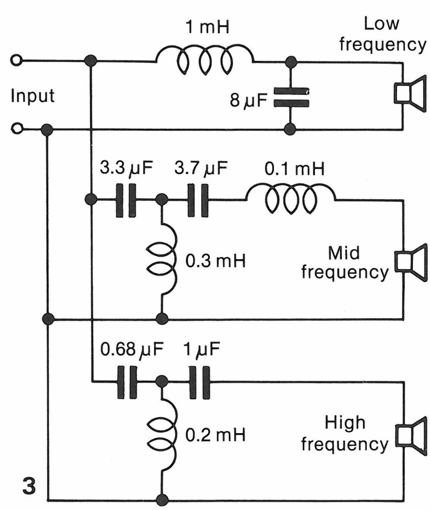
Alternative magnet assembly, suitable for an HF Unit. The diaphragm is dome shaped.



2

Typical crossover unit.
Various circuits can be used giving various degrees of steepness of cut off. First order filters give 6 dB per octave attenuation, second order, 12 dB and third order, 18 dB. Low order filters tend to provide less phase change but produce more overlap between the loudspeaker units.



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To obtain bass response from a loudspeaker, it must be surrounded by a baffle or mounted in an enclosure.

Loudspeaker Enclosures

A loudspeaker diaphragm radiates sound waves from the front and the back, the two outputs being 180 degrees out of phase with each other (while the front is compressing the air, the rear is rarefying it). Thus, unless measures are taken to prevent it, the pressure waves will rush around from the front to the back to equalise the pressure and cancel out the sound radiation. The lower the frequency, the more time there will be for the pressure equalising action, so the less efficient will be the loudspeaker.

Baffles

Ideally each side of the loudspeaker should be working in a totally separate air space, eg in the wall between two rooms, but this is not often a practical arrangement and so a baffle is used around the loudspeaker cone. To be effective, the smallest dimension of the baffle must be at least half that of the longest wavelength it is desired to reproduce, eg 3.3 mm at 50 Hz. This is obviously impossibly large for most situations and it is much more convenient to fold the baffle into the form of a box.

Infinite baffles

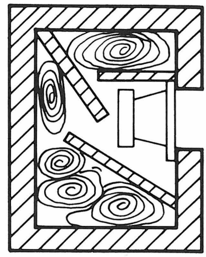
If a loudspeaker is placed in a completely enclosed, air sealed box the energy at the rear of the cone will be taken up in overcoming the stiffness of the air in the box. This will have the effect of increasing considerably the system resonance of the loudspeaker and units designed for this purpose have a very low free-air resonance. The efficiency can be improved by almost filling the box with absorbent material and by increased magnetic damping in the drive unit. Both these measures will broaden the spectrum of the resonances and make them less pronounced. The absorbtion of energy at the rear of the loudspeaker has to be paid for in loss of radiating efficiency but this is not of much importance with modern transistor amplifiers.

Vented enclosures

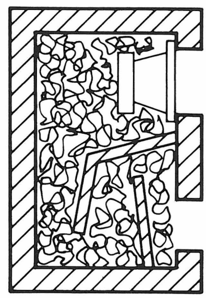
A method of obtaining extended bass response is to provide a vent in the enclosure. This effectively makes the enclosure act as an Helmholtz resonator with a resonance frequency below the cut off frequency of the unvented enclosure.

Various types of vent are used such as a small hole through the front of the cabinet or a larger hole connected to a tube to increase its length. Alternatively, an acoustic resistance or a passive unit (a diaphragm with no driving mechanism) can be incorporated in the vent. In this case the effect is to invert the phase of the radiation from the back of the loudspeaker at low frequencies so that it emerges from the box in phase with the front radiation.

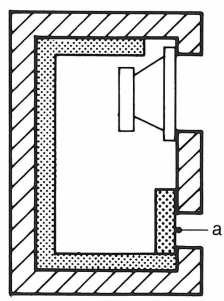
Infinite baffle enclosure containing absorbent material, eg fibreglass or rock-wool rolls, and baffles to extend the path length and reduce internal reflections.



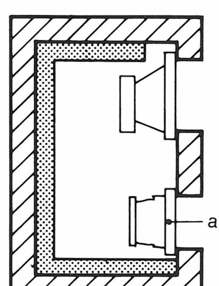
Enclosure with small vent and transmission line loading, ie the rear of the loudspeaker is terminated by a tapered duct about one eighth of a wavelength long, lined with absorbent material and terminated with a small vent.



Vented enclosure with acoustic resistance. The inside of the box is lined with absorbent material (*a*).



Loudspeaker with a large vent covered by a passive unit (*a*).



Electrostatic loudspeakers can produce excellent sound quality with a rather limited volume output. They must be carefully sited for optimum results.

Electrostatic and Plastic Film Loudspeakers

The electrostatic loudspeaker works on the principle of electrostatic attraction of unlike polarity and repulsion of like polarity. They have been manufactured with a single fixed backplate and diaphragm but these tend to have a very restricted frequency range.

The double-sided electrostatic loudspeaker

The double-sided push-pull electrostatic loudspeaker is capable of producing very high quality sound. Basically it consists of a thin conductive-coated diaphragm sandwiched between two rigid perforated backplates.

A balanced, high, polarising voltage is applied between the diaphragm and the two backplates through a high resistance. The audio signal is applied to the two backplates through a high ratio step-up transformer. The alternating potential applied by the signal causes the backplates to become alternatively of like and unlike potential with the diaphragm. The diaphragm is then attracted by one plate and repelled by the other alternatively and moves in the space between the fixed plates which are perforated to allow the resultant sound waves to radiate.

As the diaphragm approaches the fixed backplates, the spacing diminishes and the capacitance increases; this would increase the action, causing a non-linear response. This is the purpose of the high resistance in the polarising circuit. It is effectively in the charge circuit of the capacitor and, by giving it a high value, the time constant can be made longer than the period of the lowest frequency it is required to reproduce. The charge on the plates is prevented from changing at audio frequency and the action is linear.

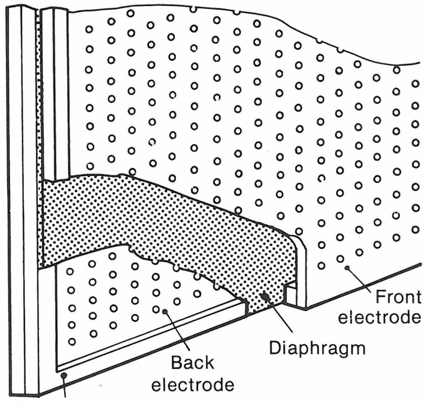
In view of the high voltage involved it is necessary to provide sufficient stiffness in the diaphragm and limit the force applied to ensure that it does not touch the backplate. As an added precaution the plates can be coated with an insulating plastic film.

In the Quad Acoustical electrostatic loudspeaker there is a separate area in the centre for high frequencies, supplied from a high pass filter. The full area radiates the bass.

Plastic film loudspeakers

The availability of thick flexible plastic film on which a conductive coating can be printed has made possible the production of some novel forms of loudspeaker which, although operating on the electrostatic principle, use the whole conducting area as a diaphragm in much the same manner as electrostatic loudspeakers.

1, Sectional view of a push-pull electrostatic loudspeaker.
The electrodes are made of plastic and only the outer surfaces are metallised.



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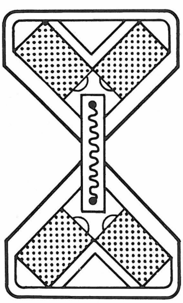

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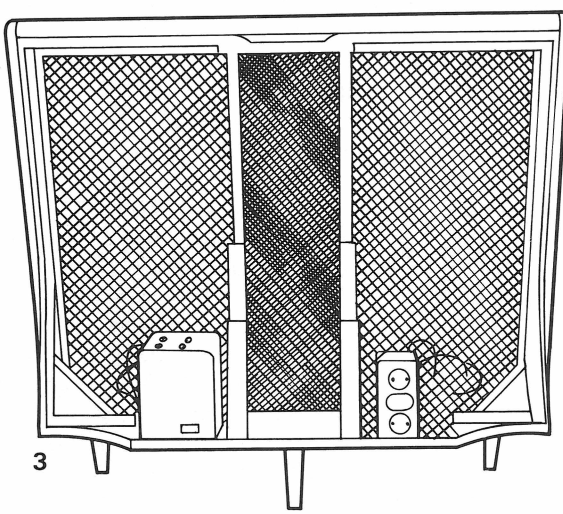


3

2, Rear view of the Quad Acoustical full-range electrostatic loudspeaker. The polar response is bidirectional at low frequencies becoming unidirectional at high frequencies.
Correct positioning of the loudspeaker within the room is important. It should not be used close to a wall.
3, Basic construction of a Heil loudspeaker driver. A corrugated conductive film element (*a*) is sandwiched between two powerful magnets (*b*) and polepiece assemblies (*c*). When an audio signal is applied to the element, the corrugations alternatively open and close in a bellows action modulating the air forced through the gap between the polepieces.

Exposure to loud sounds for lengthy periods can result in permanent deafness.

The Dangers of High Sound Levels

No book about sound recording would be complete without mention of the dangers of exposure to excessive loudness levels. Medical evidence has confirmed that long exposure to very loud sound can cause damage to the hearing mechanism, resulting in permanent deafness.

The range of human hearing

Our ears encounter an enormous range of audio power. It could vary from about 0.000 000 001 W for a soft whisper to 10 000 W for a turbo jet engine (ie 30–170 dB relative to 10^{-12} W).

To cope with this range, the human ear has a built-in limiting mechanism. When subjected to a loud sound our ears switch in a sort of attenuator so that we become partially deaf.

If the sound is not too loud or the exposure too long, our hearing recovers soon after the sound has stopped. The louder the sound and the longer the exposure to it, the longer it will take for our hearing to recover. This could take a few hours or days. In the extreme case, where exposure to loud sound has been long and not interspersed with long periods of rest, physical damage to the cochlea will result in permanent deafness.

One of the dangers of exposure to loud sounds is that the resulting deafness can come on gradually and therefore not be noticed until the condition is acute and the situation irreversible. Impairment usually starts with a dip in the 4000 Hz region which widens and deepens until there is a general loss of intelligibility.

There is a definite relationship between loudness, duration of exposure and the effect on hearing. Although it varies between individuals, the relationship between duration of sound and hearing impairment is generally linear but the effect of loudness rises as loudness increases and above 100 dB it goes up very rapidly indeed.

Whereas hitherto, exposure to very loud sounds was rare and of short duration (except for those engaged in noisy industrial processes), nowadays with the availability and popularity of high power amplifiers particularly in the 'pop' scene it is possible to listen to sound levels of over 100 dB for long periods of time.

Sound levels in discotheques often exceed 120 dB and there is evidence to suggest that many young people have permanent hearing impairment as a result.

Recording engineers engaged in recording pop music, whose clients have already damaged their own hearing and therefore demand higher and higher levels, are particularly at risk. If it is your profession, your livelihood could be at stake.

SOUND POWER AND POWER LEVEL SCALES

Power watts	*Power level dB re 10 ^{12} watt*	*Source*	
25–40 million	195	Saturn rocket	
100,000	170	Ram jet	
10,000	160	Turbo-jet engine with afterburner	
		Turbo-jet engine, 7000lb thrust	
1,000	150	4 propeller airliner	
100		4 propeller airliner	
10	140	75 piece orchestra	peak RMS levels in 0.125 sec intervals
	130	Pipe organ	
1	120	Piano	
0.1	110	Blaring radio	
0.01	100	Car on motorway	
0.001	90	Voice shouting (average long time RMS)	
0.0001	80		
0.00001	70	Voice, conversational level (average long time RMS)	
0.000001	60		
0.0000001	50		
0.00000001	40		
0.000000001	30	Voice, very soft whisper	

VALUES OF SOUND PRESSURE LEVEL IN SPECIFIC FREQUENCY BANDS WHICH INDICATE A HAZARD TO HEARING AT STATED DAILY DURATIONS FOR ONE EXPOSURE

Octave band specified as centre frequency (Hz)	*Sound pressure levels at specified durations (dB)*					
	4 hours	*2 hours*	*1 hour*	*30min*	*15min*	*7min*
63	100	103	106	110	116	122
125	94	97	100	104	110	116
250	90	93	96	100	106	112
500	87	90	93	97	103	109
1000	85	88	91	95	101	107
2000	83	86	89	93	99	105
4000	82	84	88	92	98	104
8000	81	84	87	91	97	103

Selecting Equipment

The following is intended as a guide to assist in the selection of various items of equipment used in sound recording and reproduction, with an explanation of their specification parameters and test procedures.

Microphones

When selecting a microphone for a specific purpose the primary considerations are directivity and frequency response.

Directivity. The directional response of a microphone, ie its angles of acceptance and rejection, desirable for a particular situation depends upon the acoustic environment and the need to discriminate between different sources of sound. Experience with the use of microphones shows that it is often more important to have a clearly defined 'dead' angle that can be pointed towards an unwanted source of sound (at least as far as that microphone is concerned) such as a Public Address loudspeaker, than to have a narrow axial response. In such circumstances a cardioid or figure-of-eight characteristic is indicated, the choice being determined by whether the unwanted sound is positioned towards the back or side of the microphone. If, however, it is required to discriminate between the sound source and reverberant acoustics or a surrounding noisy atmosphere, then a hypercardioid or superdirectional microphone should be selected. It is worth remembering that the 'interference tube' or 'gun' type of microphone only achieves a very narrow acceptance angle in free-field conditions (eg in the open air). In reverberant conditions where the interference tube is presented with ambient sound in random phase, there will be little better discrimination than that of the microphone to which the tube is attached.

In selecting microphones for use as a 'coincident pair' for stereophony, a well defined polar characteristic is paramount and this should be maintained throughout the full frequency spectrum. Manufacturers of high quality microphones normally provide a set of polar characteristics with their microphones showing the directional response at low, mid and high frequencies. The curves should match within a very few decibels at all frequencies.

Sometimes it is required to cover a wide area of sound at relatively close range, eg with the artists forming a circle around the microphone, or to include as much as possible of the ambient acoustics. For this purpose an omnidirectional response is ideal but it is important that this characteristic is maintained throught the frequency range at all angles of incidence to the microphone.

Frequency response. The first question to ask is whether an equal frequency response at all frequencies is really what is wanted. For the majority of applications, especially in the high quality or professional fields, the more even the frequency response the better. After all it is always possible to distort the output by subsequent 'frequency shaping'. Many of the better quality capacitor or ribbon microphones have characteristics that are flat within one or two decibels over the range 30–20 000 Hz. Even so, different models, even of the same type, tend to have their individual characteristics, notably in their response to transients (impulsive sounds). In the professional field, these small differences tend to influence the selection of microphones for each purpose.

In general, electrostatic microphones, especially those with small diaphragms, tend to have a very flat frequency response which extends well above the audio range with an excellent response to transients. In some applications this can result in giving the sound a rather 'clinical', 'edgy' character, which may be less pleasant than the softer, more rounded effect produced by ribbon or large-diaphragmed microphones.

There are some applications for which a flat frequency response is not required, notably hand-held microphones used very close to the mouth. In this case a response that 'rolls off' at about 6 dB per octave below 200 Hz is desirable to counteract the bass boost which occurs when a directional microphone is used close to a point-source of sound. It also helps to reduce handling and wind noises. On the other hand, where an accentuation of bass is required, the use of a ribbon microphone close to the mouth can give the illusion of a powerful bass voice. Some microphones, designed for voaclists' use, have a built-in switchable bass cut and a frequency response that rises in the region of 3–5 kHz. This can help to give clarity and the effect of 'presence' (closeness) to the voice and help it to cut through the accompaniment.

Handling characteristics. An important feature of a microphone, particularly one that is to be hand-held, is its robustness. In general, dynamic microphones (moving coil) tend to be the most robust and some are specially designed to stand up to rough usage in the 'pop' music field. Their internal working parts are shock mounted. Electrostatic microphones tend to be fragile due to their miniature head amplifiers and connectors and can be seriously affected by changes in temperature and humidity. Ribbon microphones are generally only suitable for static situations, being particularly suscptible to handling noise and wind. An exception to the above is the noise-cancelling 'lip-ribbon' microphone, used mainly for sports

commentaries. This has a carefully shielded ribbon element which is held at about 5 cm from the mouth by means of a spacing bar held against the upper lip. The resulting proximity effect would give the speech a considerable bass boost but this is counteracted by the design of the microphone and electrical filtering to produce a reasonably flat characteristic between 100 Hz and 5 kHz. More distant sources of noise are subjected to the equalising action, without the proximity boost, and are correspondingly reduced. Hence the term 'noise-cancelling'. It is generally not advisable to use ribbon microphones close to bass drums or high power guitar amplifiers as the large sound pressure waves they produce can distort the ribbon permanently.

Sensitivity. There is a wide variation in sensitivity between the various types of microphone and of course an enormous variation in the levels of sound pressure to which they can be subjected. This could range between picking up whispered conversation at a distance in a television or stage presentation and proximity to amplified instruments in 'pop' groups with loudspeaker outputs of enormous power.

In general the higher the sensitivity of the microphone the better as this ensures a good signal: system-noise ratio, provided that there is no risk of overload with loud material. Although it is possible to 'blast' microphone disphragms with impulsive pressure waves, the most likely cause of overload distortion is in the first amplifier. Some microphones have switched attentuators built into them or can be provided with in-line attenuators. With electrostatic microphones these usually take the form of 10 or 20 dB pads which are inserted between the microphone capsule and head amplifier.

There are various methods of specifying microphone sensitivities:

dB relative to 1 volt/dyne/cm². This CGS (Centimetre, Gram, Second) system, in which microphone sensitivities are quotes in dB relative to 1 volt/dyne/cm^2, although largely superseded, is a useful guide as this output is roughly equivalent to the level of normal speech. For example, a microphone with a sensitivity of +65 dB relative to 1 volt/dyne/cm^2 will produce an output of approximately +65 dB relative to zero level with normal speech.

dB relative to 1 volt/newton/cm². The MKS system (Metre, Kilogram, Second) units have largely superseded the CGS units. With this system sensitivities are quoted in dB relative to 1 volt/newton/m^2. To convert CGS units to MKS units subtract 20 dB. For example, +60 dB relative to 1 volt/dyne/cm^2 equals +40 dB relative to 1 volt/newton/m^2.

Open circuit voltage (mV) for 74 dB SPL. The output voltage produced by a sound pressure level of 74 dB SPL or 0.1 pascal (1 Pa = 1 newton/m^2). This is the pressure level equivalent to conversational speech at about 30 cm distance.

Output power (dB relative to 0 dB SPL). To overcome the problem of associating output voltages with the differing nominal impedances of various microphones, the power output can be specified. This is usually referred to 1 mW, the sensitivity being given in dB for a threshold-of-hearing sound pressure level of 0 dB (0.2 Pa).

TYPICAL MICROPHONE SENSITIVITIES

Type	*Decibels* CGS	MKS	*Open circuit voltage* (mVolt)	*Output power* dB rel. 0dB SPL
Dynamic cardioid	−76	−56	0.16	−150
Capacitor cardioid	−61	−41	0.95	−135
Ribbon	−85	−65	0.6	−159

Impedance. The simple crystal microphones supplied with some domestic tape recorders have very high impedance and can only be used on short leads (not much more than about 2 m) otherwise the shunting effect of the lead capacitance will result in a serious loss of high frequency response. Capacitor microphones, although inherently of very high impedance, are always used in conjunction with head amplifiers (although the capsule can in some cases be separated from it by about 1 m by means of a coazial extension tube). They usually have a nominal output impedance of 200 Ω and should work to a load impedance of at least 500 Ω.

Dynamic (moving coil) microphones and ribbon microphones, although intrinsically of low impedance, usually incorporate a transformer to bring the impedance up to 30 or 200 Ω. Most models need to work into about five times their nominal impedance.

Physical characteristics. Another important feature that governs the choice of a microphone is its size and physical appearance, as this is likely to have a bearing upon its acceptability close to a performer. In assessing this quality note must be taken of the microphone's

wind-shielding requirement. It is obviously no use selecting a vocalist's microphone for its small, neat appearance and then having to fit a large windshield to prevent pressure blasting (popping) at close range. Microphones designed for vocal applications often incorporate close-talking sheilds in their construction and do not require additional shielding for this purpose. Alternatively, small plastic foam 'close-talking' shields are obtainable, in a vareity of colours, for use with unshielded microphones. If a microphone is to be used out of doors in windy conditions, a large windshield will be required. Windshields with a diamter less than about 8 cm are not very effective in severe conditions as they can cause turbulence effects which tend to shift the noise further up the frequency scale, thereby making it more objectionable. Plastic windshields can quickly become sodden and useless in the rain. Fine plastic gauze is usually better.

Test procedures. The scientific testing of microphones requires an anechoic chamber, ie a room completely surrounded on walls, floor and ceiling with acoustically absorbent material about 1 m thick, so as to be completely without reverberation. It also involves the use of carefully calibrated sound pressure production and measuring equipment. It is, however, possible to obtain a reasonably accurate assessment by comparative listening tests.

The procedure is to set up the microphone in as dead an acoustic environment as possible (the open air is ideal if the situation is quiet and there are no reflective structures within range). If possible it should be placed alongside another microphone of known performance and arrangements made to listen to each in turn through equipment of established quality. A person whose voice is well known to the listener should then be asked to walk around the microphone at a distance of about 0.5 m continually stating his angle with respect to the microphone's main axis. His voice should be reproduced at the same level as in person. Careful note should be taken of the change of volume with angle of incidence and any changes in quality that may accompany it (disregarding as much as possible any room acoustics). The test should be repeated at different distances from the microphone and, if possible, in each case compared with the output of a known microphone in the same situation.

Placing the microphone close to a piano played fortissimo is an excellent test of high frequency and transient response. If it is possible to test the microphone in association with orchestral instruments, it is advisable to keep making comparisons between the actual sound as heard in the studio and that reproduced by a loud-

speaker at as nearly as possible the same level. It is surprising how soon one can become accustomed to a particular quality of sound and assume it to be correct when it is very far from the original.

Record players

The turntable. In choosing equipment for playing gramophone records, the first thing to decide is the degree of automation, if any, required. In the days before LP records, when the maximum playing time was about 5 min, there was much to be said for having an 'auto-changer' which would accommodate a stack of records and play them in sequence. There is much less reason to have auto-change nowadays with long-playing records but it can be useful, for example, if it is required to play a stack of 7 in singles or provide continuous background music with LPs.

A modern record-changer is a very complicated instrument. It has to be capable of dropping, one at a time, a stack of mixed 7 in, 10 in and 12 in records, locating the start of the record and sensing the end from the spiral run-in groove to trigger the pick-up to rise and the next record to be dropped in. It must detect the end of the last record and switch itself off. All this has to be achived with very light pick-ups and playing weights which allow for a maximum side pressure of the order of 0.25 g. It is done by detecting the increase in radial velocity of the tone arm as it tracks the run-out groove either by means of an electronic sensor or a mechanical trigger.

A typical mechanical system employs a small lever which engages with a projection on the turntable hub. While the record is playing the lateral movement is small and the lever is progressively pushed away on each rotation by the projection. When the larger movement (per rotation) occurs on the run-in, however, it moves too far to be pushed away, engages with the projection and actuates the mechanism. In spite of the sophistication of modern equipment, record-changers are not to be recommended for the serious listener as they tend to maltreat the discs; there is a risk of deformation of the discs and spindle holes and trapping of dust and dirt between them as they are dropped on to each other.

Semi automatic reproducers. A degree of automation can be useful for single-play reproducers to drop the pick-up neatly on the beginning of the record and remove it at the end with much less risk of damage to record or stylus than by hand. The Accutrack system takes this a stage further and provides the facility to select individual bands of records. It does this by detecting the blank bands by reflecting a beam of infra-red light between a source and

detector mounted on the pick-up. (The grooves scatter the light and reduce the reflection.) The device counts the tracks as it scans across them and can be programmed to select and play a particular one.

Turntable drive systems. There are three basic systems of driving the turntable in common use.

1. Peripheral drive. A stepped-diamter shaft on the motor spindle (different diameters for different speeds) drives a rubber idler wheel which engages with the inside of the turntable platter. This system is found on many of the cheaper models. It can suffer from 'flats' developing on the idler wheel and slippage in the drive due to dirt and oil causing flutter or wow.
2. Belt drive. A belt is used to transfer the drive from a stepped pulley on the motor spindle to the periphery of an inner turntable mounted under the platter. This system provides a very positive drive and can produce excellent wow and flutter figures, provided that the belt and pulleys are kept in good order.
3. Direct drive. The turntable is driven directly on the spindle of a slow running multi-pole motor. Speed change can be accomplished by changing the drive frequency or by switching the number of poles in use. Most examples of this system are speed-controlled by a servo system governed by a quartz-locked oscillator. They can achieve a very high degree of speed, accuracy and freedom from wow and flutter. They are mechanically simple and robust.

Turntable specifications. The important features of a turntable are speed accuracy, stability, freedom from rumble (low-frequency vibration) and, for some applications, time taken to get up to speed.

Accuracy of speed can be checked with a stroboscope, either built-in or in the form of a card placed on the platter and viewed by fluorescent light. Once up to speed, there should be little or no variation over a period of several hours. Wow (slow cyclic variations of speed) or flutter (speed variations with a repetition rate above about 10 Hz) should be better than 0.2%. If a wow and flutter meter is not available, a reasonable judgement can be made by ear (which after all is the ultimate criterion). Choose a record with long sustained pure notes in the middle of the audio range (organ music can be ideal) and listen for a 'souring' of the notes to indicate wow. Take note of any tendency of the turntable to slow down with heavy passages of modulation. Similarly a solo passage on the flute or sustained note on the violin will illustrate flutter, which will produce a 'roughening' of the sound.

A record player for professional purposes, where accurate cueing

is required, should come up to full speed in a quarter of a revolution. If the record player is to be used on anything but a very solid surface or in proximity to a loudspeaker it should have effective shock-mounting between the deck and the case. Place the pick-up on a record with the turntable stationary to check for this.

To check for rumble (low frequency vibration) find a record with a long run-in groove or period of very low modulation and briefly turn the volume and bass control up high. Some test records have an unmodulated section for this purpose. There is also a device which fits on the turntable spindle and allows it to rotate while the stylus rests on a stationary 'platform' for the measurement of feed-back and rumble.

Pick-ups

There are various basic types of pick-up and it is important to choose the type most suited to the particular circumstances. The main types of pick-up available nowadays are: piezo-electric (crystal or ceramic) or electromagnetic (moving-iron, moving-coil, moving-magnet, induced-magnet and variable reluctance).

Crystal and ceramic pick-ups have a high output, commonly of the order of 0.5 volts and practically independent of frequency, so that they do not require equalisation or high amplification. This type of pick-up is normally connected to the auxiliary input of an amplifier. Crystal pick-ups have certain disadvantages which reduce their suitability for very high quality applications. Due to the method of their construction, it is difficult to achieve the high degree of vertical as well as horizontal compliance necessary for good stereo performance. Crystals tend to be affected by changes in temperature and humidity. They have a very high impedance so that the lead to the first amplifier must be very short and with low capacity, otherwise serious loss of high frequency response will result.

Magnetic pick-ups. Magnetic pick-ups have lower outputs and lower output impedances than the above types being of the order of 5 mvolt and 50 kΩ. As their mode of operation is electromagnetic, their output tends to be proportional to rate of change (ie velocity of vibration). In other words, the output would tend to rise as frequency is increased in the order of 6 dB per octave, so reciprocal equalisation is required to produce a flat characteristic. In fact recording/reproducing characteristics are agreed by international standard and should follow a curve in which the slope varies with frequency as a combination of three curves which, for fine-groove recording have time constants of 75 μsec, 318 μsec and 3180 μsec.

Magnetic pick-ups are capable of producing very high quality

but a high degree of amplification is required and the most stringent precautions must be taken to avoid the pick-up of stray magnetic fields such as mains hum (possibly from the turntable motor). This is particularly true of moving-coil pick-ups which tend to have very low outputs and impedances (typically 0.02 mvolt and 5 Ω).

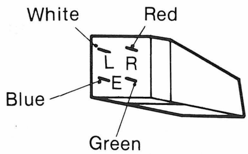
Magnetic pick-ups can be supplied with a transformer to make them interchangeable with other magnetic cartridges but great care must be taken in positioning the transformer to avoid hum pick-up.

General characteristcs. The frequency response for a high-quality cartridge should be within 2 dB between 20 and 20 000 Hz with a balance between the two stereo channels within 2 dB and separation about 30 dB at 1 kHz and 20/25 dB at 10 kHz. Apart from the above, the most important feature of a pick-up is its 'trackability'. This is a function of the mass of the stylus and its compliance (the ease with which it moves to follow the contours of the groove). Generally speaking the lower the mass of the stylus and the greater its compliance, the better will be the reproduction and the less the record wear. Compliance is generally quoted in millionths of a centimetre per dyne of applied force. A good figure would be 30–50 (×10–5 cm/dyne).

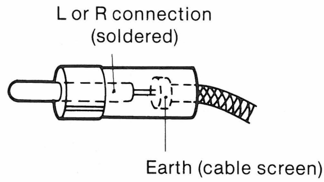
Tone arms. It must be remembered that low mass/high compliance pick-up cartridges require a very light tracking weight; a stylus pressure of about 1 g is common. These can only be played with a high quality tracking arm with minimal bearing friction in both directions, adjustable balance and bias compensation. It is pointless to try to use a high quality cartridge on a poor tone arm. The tone arm and pick-up form a combination with a natural resonant frequency and, ideally, they should be matched to each other. Bias compensation usually takes the form of a small weight attached to a string or a light spring which pulls the countereight end of the arm towards the turntable.

When using high quality light-weight pick-ups, it is essential that the turntable and mounting board are absolutely level and free from external vibration.

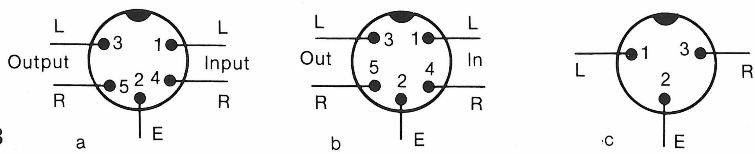
Checking pick-up tracking. To check trackability, first ensure that the turntable is level and the tone arm properly set up as regards bias compensation etc. Play a special trackability test record or recording with high modulation and smooth tone. Progressively reduce the tracking weight until distortion is perceptible on heavy passages (due to the stylus riding up the grooves). If the system is in good order this weight should be slightly less than the recommended tracking weight for the cartridge. One type of test signal consists of



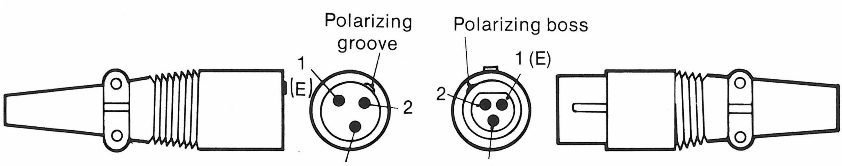
1 Stereo pickup cartridge pin connections



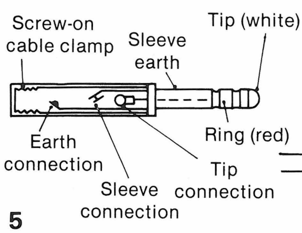
2 Phono Plug



3



4 XLR-3-12C Plug XLR 3-11C Socket



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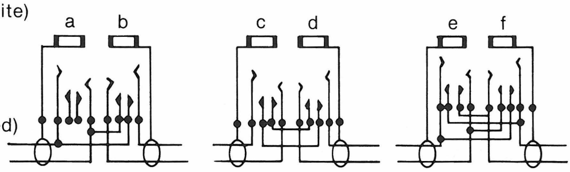

Half Normalled Single normalled Double normalled

6

1, Stereo pickup cartridge pin connections. 2, Phono plug. Screened wire should be used for all low-level audio wiring. The screen should be connected to earth at one point only to prevent the formation of an earth loop and the consequent 'hum' problem.

3, Examples of DIN socket connections. Type *a* is the accepted standard for stereo input/output connections. Pin 2 is normally earthed.

4, The XLR3-12C plug and XLR3-11C socket. Pin 1 is normally connected to the cable screen.

5, Jack plug. 6, Jackfield connections. Listening facilities are provided by *a. b, c* and *d* are break jacks. *e* and *f* provide connection in both directions but the through connection is only broken if plugs are inserted in both.

short bursts of high modulation high frequency. Bad tracking causes an unequal waveform which is manifest as a low frequency signal at the burst repetition rate.

The stylus. Most styli are either sapphire or diamond. Diamond is the more expensive but tends to have about 10 times the life of sapphire if it is carefully used (about 2000 hours). The stylus tip must be shaped so that it traces the grooves accurately without actually touching the bottom (which tends to be rough and contain dirt). Many of the better cartridges employ an elliptical-shaped stylus (the broader dimension being at right angles to the groove) so that they can 'read' the fine high frequency contours while having a broad enough profile to prevent them touching the bottom of the groove. There are also 'multi-radial' styli with a more complex shape developed to meet the stringent high frequency requirements of CD-4 four-channel records.

Tape recorders

The first question to decide is whether to go for a reel-to-reel, cassette or cartridge tape recorder. Each has its own particular advantage and suitability for particular purposes.

Reel to reel machines. The main advantages of reel-to-reel machines are:

1. Very high quality.
2. Ease of editing.
3. Robustness.
4. Long playing times (a 10 in spool of LP tape can provide up to 6 hours of continuous play at 4–68 cm/sec).

Some models are reversible, thereby doubling the playing time and can be arranged to repeat the sequence continuously.

The main disadvantages of reel-to-reel tape recorders are:

1. Awkwardness in threading the tapes.
2. Tapes can be large and cumbersome to store and identify.
3. High quality machines tend to be large, heavy and expensive.

Cassette machines. The main advantages of cassettes are:

1. Convenience and ease of operation.
2. Tapes are compact and easy to store and identify.
3. Machines can be small and compact and generally reasonably inexpensive.

Cassette machines can be capable of excellent quality but this is only achieved by exploiting the extremes of the technology. Only a few years ago, it would have been considered impossible to obtain full frequency range reproduction with tape running at 4.8 cm/sec

with a track width of only 0.26 mm. Nowadays this can be achieved but only with a well adjusted and maintained machine using good tapes and the correct bias settings etc.

Editing can be a problem with cassettes. Although it is possible to edit cassette tapes by physical cutting and splicing, it is a very delicate operation and virtually impossible to obtain the same accuracy as with reel-to-reel, largely due to the difficulty of accurately locating and marking the edit point. It is, however, possible to edit by dubbing from machine to machine using the pause control.

'Dolby' noise reduction is essential to obtain good signal:noise ratio.

One of the main disadvantages of magnetic recording as compared with the disc has always been the difficulty of selecting a specific item in a recording. This has now been overcome in some of the most sophisticated machines by the provision of a 'search' facility. The various functions of the machine are controlled by a logic system. This can be 'programmed' to find a specific item by counting the pauses between the recordings while the tape is being spooled (the heads being engaged with the tape for the purpose) and comparing the number with a preset figure. It can also be programmed to repeat a recording from a preset zero on the tape counter.

The search facility can be very useful but its use should be restricted to prevent undue wear on the tape heads.

Cartridge machines. There are basically two types of cartridge, the eight-track long-play used mainly for in-car entertainment or background music and the professional machines used for accurate cueing-in of announcements, 'jingles' and sound effects etc., or for the continuous playing of background music where quality is important and repetition times can be relatively short. (The large 'C' type cartridges can contain a loop with about 30 min duration at 19 cm/sec).

Specification. The qualities to aim for when choosing a tape recorder are:

	Reel-to-reel 19 cm/sec and professional cartridge	*Cassette*
Speed accuracy	Within ±0.2% Preferably adjustable	±0.5%
Speed stability *Wow and flutter*	DIN (peak value weighted) 0.1% WRMS 0.05% (NAB)	0.2% 0.06% (NAB)

continued on next page

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	Reel-to-reel 19 cm/sec and professional cartridge	*Cassette*
Tape slip	Less than 0.2%	Less than 0.3%
Tape timer accuracy	±1%	±2%
Frequency response	30 Hz–15 kHz ±2 dB 60 Hz–12 kHz ±1 dB	With Dolby off: 30 Hz–17 kHz ±3 dB (with FeCr Cassette)
Total harmonic distortion	Tape flux 320 nWb/m 1% or less Tape flux 525 nWb/m 2% or less	1% with FeCr cassette
Spooling	Spooling should be fast, the wind smooth, leaving the edges of the tape even. One test is to disconnect the mains supply during fast spooling. A good machine should halt the tape without damaging it.	Machines should have auto stop. Some machines have automatic search facilities. This allows spooling with the tape in contact with the replay head. They can be programmed to identify and count the pauses between items and select the required one.
Metering	The provision of effective level indicators is important to achieve the correct level of recording. These can be in the form of meters or light emitting indicators. In general peak programme indication is preferable to VU metering as it gives a better indication of possible overload. VU meters give an indication of average power.	

Choosing tape. There is a wide variety of tapes available nowadays. Probably the best advice is to obtain the best that one's equipment is capable of handling. Really cheap tape can be a poor economy because, apart from its limited frequency response, it tends to shed oxide and clog head gaps. Chrome dioxide tape has a superior frequency range to standard tape but it requires a higher level of bias and different equalisation which is not available on all machines. It also tends to increase head wear although some manufacturers incorporate means of reducing this. Ferrite heads with fused glass

gap fillers can provide long life even using the more abrasive types of tape. Metal tape (tape using pure iron instead of oxide) can provide very wide frequency response and exceptional dynamic range but it requires powerful bias, heads capable of accepting the high levels without saturation and exceptionally powerful erasure, due to the high magnetic retention (coercivity) of the tape.

Selecting amplifiers

Audio amplifiers can be broadly classified into pre-amplifiers, power amplifiers, integrated amplifiers (pre-amp and power amp combined) or functional amplifiers (combined with reproducing equipment, usually a radio tuner).

The advantages of having a separate pre-amp include: the possibility of a more versatile unit capable of accepting a wide variety of different types of input with suitable equalisation; less chance of disabling the entire equipment in the event of failure; and the ability to interpose equipment such as graphic equalisers, reverberation units and noise reduction systems between the two amplifiers (some integrated amplifiers provide the means of breaking the connection between the two units for this purpose). Separating the power amplifier could enable it to be sited for maximum heat dissipation. Some loudspeakers have built-in amplifiers, a technique that is particularly suited to active crossovers and motional feedback systems (see p. 187).

Facilities. The facilities available on a pre-amplifier or integrated amplifier should include bass and treble equalisation with possibly an adjustable 'presence' peak centred on about 3 kHz. Extreme bass and treble roll-off filters should be provided, either on a switchable or permanent basis. Direct-coupled solid-state amplifiers are capable of reproducing a range of frequencies from 0 Hz (DC) to over 100 kHz and there is a risk of subsonic frequencies caused, for instance, by a warped record intermodulating with the audio frequencies to cause distortion. There is also the possibility that a sharp sound, caused perhaps by dropping the pick-up on the record or a momentary disconnection, could cause the loudspeaker cones to jump out of their gaps. A fairly sharp rool-off of at least 6 dB per octave is required below about 20 Hz.

High frequency attentuation can be used to reduce record scratch and system or tape hiss. A 'loudness' control should be available to boost the treble and bass to compensate for the relative insensitivity of our hearing when the volume is low (see p. 12). A balance control (with click stop in the centre) is necessary. Stereo/mono and phase-reversing switches can be useful, especially when setting-up the system.

Input switching is required for phono (gramophone) with equalisation to follow the RIAA (Record Industries Association of America) characteristic for magnetic pick-ups. A high-gain microphone connection (two for stereo) should be provided, preferably with a means of mixing its output with another source. Auxiliary and tape playback inputs should be provided with a straight characteristic (tape equalisation is normally contained in the tape recorder). A line-level output should be available to feed a tape recorder. Outputs should be provided to feed a set, or preferably two sets of loudspeakers, with a means of switching them. A stereo headphone jack should also be provided.

Typical amplifier specifications. The power output required of an amplifier depends upon the efficiency of the loudspeakers, the size of the room and the type of programme material for which it is to be used. High quality music reproduction in a typical domestic situation requires an available power output of between 10 and 20 W for average music reproduction. On the other hand an amplifier used for a bass guitar or to provide music in a 'disco' situation would need a very much higher power output. The modern trend to reduce the size of loudspeakers by using small large-movement units in infinite-baffle cabinets increases the power requirement of the amplifiers considerably.

The actual power that an amplifier delivers to a loudspeaker depends upon its motional impedance and its relative resistive and reactive values at various frequencies. It is normal, therefore, to rate amplifiers for their 'steady-state' power, dissipated into a resistive load equal to the nominal load of the loudspeaker to which it is to be connected. The power rating is for a given frequency range and within given distortion parameters. Sometimes 'peak programme' or 'music power' ratings are quoted. Due to the intermittent nature of most programme material much higher figures can be quoted for music power than for 'continuous RMS power'. In the case of a pure sine wave RMS power would be half peak power but the 'envelope' of music waveform is much more complicated and the relationship between the two systems of measurement varies with the type of programme material.

Two factors in particular affect the power rating of a transistor amplifier:

1. The risk of overheating and 'thermal runaway' when sustained power is called for.
2. The capacity of the power unit.

Small capacity power units can supply high current for short periods but may be incapable of sustaining it for long. In this

context the demands made by classical music, where the periods of high modulation tend to be short lived, can be much less than those of 'pop' music, where the material is highly compressed and large power output is required to drive the relentless bass (often required to be reproduced at high volume through small, inefficient loudspeakers).

Figures for amplifier distortion are often quoted for maximum power output but this does not necessarily tell the whole story. Transistor amplifying characteristics tend to follow an expotential curve, with pronounced origin curvature, which can create crossover distortion at low level as the signal changes polarity between the two elements which are combined to form a class B output stage. This can take the form of particularly objectionable harmonics affecting quiet musical passages and speech which can sound worse than the measured distortion value would suggest. It is, therefore, desirable to ascertain the total harmonic distortion values at low as well as high levels of output.

Typical specifications for a good quality domestic stereo amplifier could be:

Power output	30 W continuous RMS for 20 Hz–20 kHz within rated harmonic distortion.
Total harmonic distortion	Less than 0.2% at all levels up to rated maximum rated output.
Input sensitivity	Phono 3 mV into 47 kΩ, with signal:noise ratio of 65 dB. Auxiliary 150 mV into 47 kΩ, with signal:noise ratio of 70 dB.
Overload level	Phono 300 mV RMS. Auxiliary 9 volt.
Tone controls	±8 db treble and bass lift or cut. Preferably also a middle 'presence' control and extreme bass and treble filters.
Loudness control	Increase in bass and treble at low volume settings up to +8 dB at 50 Hz, +6 dB at 10 kHz.
Switching	Input switching for phono, aux, tape and radio. Output switching for one or two sets of stereo loudspeakers and headphones. Mains outlets for anciliary equipment (switched).
Stereo balance	Preferably with centre click stop.
Hum and noise	Phono 75 dB. Aux 90 dB. Unweighted signal: noise ratio with reference to rated output power.

Tuner amplifiers

It is quite common nowadays to combine a radio tuner with an integrated amplifier, thereby saving cost and possible interconnec-

tion problems, by employing one power unit to supply both functions. There is much to be said for this technique, especially as most combined systems available today provide most of the facilities normally found in individual units. In choosing a tuner the first two points to be decided are: what wavelengths are required and whether preselected tuning (preset push buttons) is desired.

Most tuners provide for the medium wave band with amplitude modulation (the 'carrier' wave to which the set is tuned is varied in strength by the amplitude of the audio waveform) and VHF band with frequency modulation (the frequency of the carrier wave is varied by the audio waveform) which also carries the stereo signal. VHF, FM is essential for good quality listening.

There are comparatively few HiFi tuners that also provide short wave bands.

If the tuner is to be used for regular listening, especially if it is to be operated by all members of the family, there is much to be said for preset push-button tuning to remove the hassle of finding the correct tuning position each time the station is changed. Tuners should incorporate visual tuning indicators and it is normal to incorporate a lamp which lights when the stereo pilot tone is being received at sufficient level for stereo reception.

Typical High-quality Receiver Specifications

Medium wave

Frequency range:	522 Hz–10 kHz (amplitude modulation)
Sensitivity:	Better than 20 μvolt for 10 dB signal:noise ratio
	Better than 100 μvolt for 40 dB signal:noise ratio
Frequency response:	Within 3 dB, 20 Hz–16 kHz
Harmonic distortion:	0.5% at 400 Hz

Frequency range:	88 mHz–108 MHz. Frequency modulated, stereo
Sensitivity:	*Mono* 3.5 μvolt for 50 dB quieting (ie signals below this threshold are suppressed to reduce intersignal noise)
	Stereo 45 μvolt for acceptable stereo signal
Signal: noise ratio	75 dB (mono), 70 dB (stereo)
Limiting characteristic:	Limiting commences at input levels above 2 μvolt. (Amplitude limiting reduces the

	possibility of fading and interference in FM signals)
Frequency response:	±3 db from 30 Hz to 20 kHz (allowing for the standard de-emphasis characteristic of 50 μsec in Europe 75 μsec in America, 25 μsec with Dolby NR)
Harmonic distortion:	0.1% (mono), 0.3% (stereo)
Capture ratio:	2 dB. This is the so-called 'capture effect' of frequency modulation. When two closely spaced signals are received together the resultant AF output contains both FM and AM components. The limiting action of the receiver removes the AM component and the stronger FM signal takes command, suppressing the weaker one, thereby reducing the risk of co-channel interference.

The above sensitivity figures assume the provision of a suitable antenna for each waveband. This is always to be recommended. It is also recommended that when an external antenna is used a suitable lightning arrestor is provided to prevent the risk of damage to input transistors, although in some cases the series inductance of the down-lead is sufficient to guard against this where a long coaxial lead is employed.

Cassette recorder tuner amplifiers

An extension of the above arrangement incoporates a cassette tape recorder with a tuner amplifier. Some examples of this arrangement are very compact and yet provide more than adequate power to feed loudspeakers.

This type of equipment is especially convenient for recording broadcast programmes, especially if a suitable 'pause' control is provided on the record function to eliminate announcements and commercials. Some models have built-in timers which can be set to switch the machine to record programmes on a time cue while unattended. This can be a very useful facility, as there is then no need to miss one's favourite programme through absence or competition from the television.

Music systems (music centres)

Finally, there is the composite music system or 'music centre', a unit which combines tuner/amplifier, cassette machine and record player. Early examples of this type tended to be of rather poor specification and limited power output and were often rather dis-

paragingly compared with the old 'radiograms' because they incorporated several functions in one case. Even the cheaper music centres, however, have two major improvements: the provision of a cassette machine and the fact that the loudspeakers are separate and can be placed in a room for convenience and, hopefully, for stereophonic effect.

In many of the middle-range units the loudspeakers are, in fact, the weakest link in the chain and owners of this type of equipment would probably find it worthwhile to obtain a really good pair of loudspeakers. (The existing ones could then be added for 'surround sound' listening for which the frequency-range requirements are not quite so stringent. See p. 52.) On the other hand, in the case of the cheapest range of equipment, the provision of wide-range loudspeakers could show up some unpleasant characteristics such as turntable rumble.

Recently a new range of high-quality music systems has become available. These combine all the advantages of the basic music centres in terms of compactness and the interconnection of facilities with standards of reproduction that compare with high-class individual units. Moreover, the provision of logic control for the various functions not only enables them to be operated by touch switches but also enables them to be interlocked. It is thus possible, for example, for the radio and tape recorder to be controlled by a built-in time switch and programmed to turn on and record a broadcast at a specific time. Similarly the cassette recorder can be linked to the pick-up so that recording from disc starts automatically when the pick-up is dropped on the record and stops when it is lifted at the end.

The use of logic control can also enable the 'search' facility whereby a cassette recorder, or pick-up, can be programmed to find a particular band of recording, in the cassette machine by counting the gaps between the recordings and with discs (using a tengential tone arm) by counting the spaces between the 'cuts'.

The important questions regarding composite music systems are whether one needs to obtain a complete set of facilities simultaneously and whether the individual specification for each piece of equipment is ideal. There is also the matter of putting 'all the eggs in one basket' with the consequent risk that a failure in one section could disable the whole equipment. Obviously the arrangement can be very compact and there is a saving in overall cost due to the sharing of the power unit and cabinet etc. but the high 'packing density' of the equipment can create problems for servicing. The very rapid development in audio techniques must also be taken into account. The advent of digital techniques is already pointing towards the obsolescence of analogue disc reproducers and this could soon be true also of magnetic recorders. It would be a pity if the high capital outlay involved in a high quality music centre should inhibit the

owner from following up the magnificent advances that digital techniques are producing in the field of audio.

Copyright considerations

It should be emphasised that the recording of broadcast material, especially if it involves music or scripted speech, is almost certainly an infringement of copyright, both as regards the artists or recording company concerned and the broadcasting authority.

Broadcasting organisations tend to turn a 'blind eye' to domestic recording for personal use but the reproduction of copyright material for sale or public performance is a serious offence unless the permission of the holder of the copyright or their agent (in UK usually the Mechanical Copyright Protection Society) has been obtained and a royalty paid (typically $6\frac{1}{4}\%$ for each copy).

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Glossary

Absorption coefficient (24) The ratio of acoustic energy absorbed by a surface exposed to a sound field to that incident on the surface. It is equal to 1 minus the reflection coefficient of the material.
Acoustic effect (22) The effect of the surrounding environment on the sound.
Acoustic treatment (24) The application of a material to the surface of an enclosure to modify the acoustic.
Ambient sound (18, 44) The noise, reverberation or atmospheric sounds that form a background to the principal source.
Amplifier (56, 136) A device with which strength of a signal can be modified. Can also be used for changing impedance for matching purposes.
Amplitude (12) Peak value of a waveform.
Attack time (60) The time taken for the gain reduction to take effect in a compressor or limiter.
Attenuator (24, 56) That which reduces the level of a source. In the electrical sense it could be an impedance. In acoustics it might be absorption introduced by acoustic treatment.
Audio (12) Within the normal frequency range of human hearing (about 20–20 000 Hz).
Azimuth (102) The angle between the gap of a tape head and the longitudinal axis of the tape.

Back tracking A method of building up a sound recording in which the artists add successive parts to a previously recorded track.
Backing Accompaniment.
Backing track Pre-recorded accompaniment to which a vocal or solo part is added.
Balance (58) The achievement of the correct relationship between the various components of a sound source and the acoustic environment to produce the optimum artistic effect.
Bass (14) The lower notes in the audio frequency range.
Bias (92–98) In magnetic recording, the superimposition of a steady magnetic field on the programme signal magnetic field to bring the signal excursions to a straight part of the magnetisation characteristics and thereby avoid distortion. Bias can be DC or, more usually, high frequency AC (of the order of 100 kHz).
Bias compensation (148) In pickups, a small torque, usually applied by a weight on a string, to compensate for the inward force caused by the drag of the pickup on the record acting on the effect angle of the pickup arm.
Binaural hearing (46) The ability to perceive the direction of a sound source, either directly or by electroacoustic means whereby the two ears are supplied by separate transmission channels. Micro-

phones for binaural sound are usually embodied in a dummy head to simulate the natural condition.

Capacitor A component made up of conducting plates separated by insulation (termed dielectric). Capacitors present a decreasing impedance to alternating current as the frequency is increased. They do not pass DC.

Capacitor (or electrostatic) microphone (36) A microphone in which the sound varies the capacitance between the diaphragm/s and a fixed back plate. The electrical output is generated across a resistance in series with the capacitor and a polarising voltage.

Capstan (116) The motor-driven spindle in a tape recorder that drives the tape at the required speed. The tape is pressed against the capstan by the pinchwheel.

Cardioid microphone (31) A microphone with a heart-shaped directivity pattern.

Cartridge (disc) (152–160) The disc reproducing head that incorporates the stylus and fits on the end of the tracking arm.

Cartridge (tape) (128–130) A box containing a single spool which dispenses a continuous loop of 6.3 mm lubricated tape, which is fed from the inside of the spool and returned to the outside. The spool is not driven; movement of the tape is caused by contact with a capstan and pinch-wheel when the cartridge is inserted in the player. There are two basic types of cartridge: the domestic type which has eight tracks available as four stereo pairs and normally runs at 9.5 cm/sec; and, the professional cartridge players which run at 19.05 cm/sec. The tape has two/three tracks, one/two of which contain the mono/stereo programme and the other the cueing information in the form of tones which can be used to stop the machine (and possibly to start another) precisely at a cue point on the tape. The standard cartridge measures $101 \times 133 \times 22$ mm and there are two larger sizes for extended play time $152 \times 178 \times 22$ mm for up to 16 min at 19.05 cm/sec and $194 \times 216 \times 22$ mm for up to 32 min at 19.05 cm/sec.

Cassette (122–124) Usually the Philips Compact Cassette format—a reel-to-reel system in which both the feed and take-up spools are incorporated in a plastic box. The 3.81 mm wide tape is driven at 4.76 cm/sec by an external capstan and pinch-wheel. Four tracks are available for stereo cassette operation—one pair is used for each direction. As the stereo pair are adjacent, occupying half the tape width, mono and stereo recordings are compatible.

CCIR (110) An international standard of the Consultatif International Radio Consultative Committee.

Chromium dioxide (CrO_2) (86) A type of magnetic tape coating which permits the recording of higher levels at high frequencies than ferric oxide but requires a different value of bias.

Coincident pair (48) An arrangement of microphones for stereophony in which two microphones are placed with their diaphragms pointing in different directions but so close together (possibly in the

same case) that the path length between the sound source and the two microphones is almost the same.

Coercivity (78) The ability of a magnetic material (eg tape) to retain magnetism, ie resistance to self or induced demagnetisation.

Coloration (14) A form of distortion of an audio signal caused by the addition of spurious harmonics.

Companding (66) A method of applying compression to a signal prior to a process (eg recording) and a complementary expansion afterwards. The result is a reduction of the system noise in the final output.

Compressor (60, 70) A variable gain amplifier in which the gain is controlled by the input signal. It is used for reducing the dynamic range of a signal.

Cross-field bias (96) A method of applying bias through the back of a recording tape by means of an auxiliary head carrying the bias waveform only. The purpose is to cause a rapid reduction of the bias waveform following the point where the tape leaves the audio recording head gap to reduce the de-magnetising effect of the bias. It also ensures that high level signals, which penetrate the tape more deeply, receive more bias.

Crossover frequency (186) The frequency at which the diversion takes place in a dividing network, eg the frequency at which the 'tweeter' takes over from the 'woofer' in a loudspeaker.

Crystal (piezo-electric) microphone (32) A microphone in which the output is generated by the sound field distorting a wafer of crystal usually of rochelle salts, quartz or tourmaline. These crystals have the ability to generate small voltages when physically stressed.

Cycle (12) One complete sequence of a variation which occurs in a periodic manner.

Damped membrane A type of acoustic treatment in which sound energy (usually the lower frequencies) is absorbed by causing an inert, highly damped surface to vibrate.

DBX (70) A noise reduction system which employs pre-emphasised compression and de-emphasised expansion.

Dead acoustics An area where there is very little reverberation.

Decibel(dB) (12) A measure of the relative intensity of sound. The decibel scale corresponds to a logarithmic law, as does human hearing, in relating sound intensity to the sensation of loudness. The original unit was the bel but this is too large for most applications so the decibel (one tenth of a bel) is used. One decibel represents about the smallest change of sound level that can be detected by the ear on steady tone. On a programme, a change of 2 dB is just discernible. The decibel is a unit of ratio between two powers or pressures. If it is required to use a decibel scale to denote an actual value, it is necessary to establish a reference with which it can be compared. To describe the actual loudness of a sound it is usual to

quote its sound pressure level measured in dBs above a standard pressure level of 2×10^5 N/m^2.
dBA When describing the loudness of a sound in relation to its sound pressure level, it is necessary to take into account the unequal sensitivity of the ear to sounds of different frequencies and intensities (as shown by the contours of equal loudness after Fletcher and Munson or, later, Robinson and Dadson). Sound pressure level is, therefore, usually measured through a 'weighting' network which roughly corresponds to the equal loudness contour. Such measurements are termed 'A weighted' and the units are dBA.
De-emphasis A response that decreases with frequency.
Diffraction (13) The manner in which sounds are able to bend around obstacles with dimensions smaller than the wavelength of the sound.
Diffusion (20) The distribution of reflective paths for sound waves within an enclosure.
Digital sound (172) A process of sound transmission in which the normal analogue waveform is converted to a series of numerical measurements which can be described and transmitted as a digital code. Usually a binary code is used so that the receiving equipment only has to recognise two alternative conditions, *on* (above 50%) and *off* (below 50%) so the system is very robust and distortion free. This is, however, achieved at the expense of bandwidth.
Dolby (66) A noise reduction system named after its inventor Dr Ray Dolby. It is a frequency-selective companding arrangement. The signals are compressed and the level increased before recording or transmission and consequently expanded. The volume (and with it the noise) is reduced afterwards.
Dolby A (66) Dolby A is a system intended for professional use, principally in the production of master tapes. It divides the audio spectrum into four bands and processes each band separately, thereby eliminating the 'breathing' effects that can occur when sound of one frequency controls another. It makes full use of the masking effect whereby our hearing is insensitive to sound of a particular frequency when simultaneously exposed to a louder sound of similar frequency.
Dolby B (68) Dolby B is a system mainly used for commercial recordings (particularly cassettes). The process is applied to the recording and a corresponding process is incorporated in the reproducer. The signal passes through a 19 kHz low-pass filter and then divides, one path going direct to an adder and the other going to the adder via a variable high-pass filter and an amplifier. The cut-off frequency of the high-pass filter is controlled by the signal level so that it reduces as the level is reduced. When the input to the equipment is high enough to mask the high frequency noise the filter passes only extreme high frequencies and the effect on the output is small with most of the signal being supplied by the direct path.

Dropout Momentary loss of signal due to localised loss or fault in tape coating.
Dubbing The process of re-recording from one recording to another. The term is also used for the addition of sound, eg dialogue to a previously recorded picture.
Ducking A method of using a compressor in which the volume of one signal is controlled by another. A typical example is when a commentary is made to 'duck' background music whenever the announcer speaks.

Echo Discrete, separately identifiable repetitions of a sound due to reflections from hard surfaces. This term is sometimes used erroneously to represent reverberation.
Eigentones (20) Standing-wave resonances set up in an enclosure when the reflective path lengths correspond to the wavelength of a sound.
Electret diaphragm (36) A diaphragm used for electrostatic microphones which has been given a permanent electrostatic charge, thereby eliminating the need for polarising supplies.
Electrostatic microphone (36) A microphone that operates by virtue of variations in capacitance between the diaphragm and the back-plate spaced closely behind it.
Erasure (88) The process of removing previously recorded signals from magnetic materials prior to recording.
Extinction frequency (90) The frequency at which signal cancellation occurs in magnetic recording owing to the width of the reproducing head gap being equal to the recorded wavelength. The extinction frequency defines the upper frequency limit of the direct magnetic recording processes. The lower frequency limit is also a function of head-gap size. The output is proportional to rate of change of magnetic flux across the gap which reduces (at the rate of 6 dB per octave) as frequency is reduced and the wavelength becomes long in relation to the head-gap. The normal maximum range for direct magnetic recording is therefore limited to about 10 octaves.

Feedback A proportion of the output of an amplifying system that is returned to the input.
Flutter Periodic variations in pitch of a recording with a fluctuation frequency above 10 Hz.
Flux density (74) A measure of the concentration of a magnetic field in lines/cm^2, Gauss, or Webers per metre of tape width.
Frequency (12) The rate of repetition of a periodic function measured in cycles-per-second or hertz (Hz).

Graphic equaliser (56) A frequency-response shaping filter in which the spectrum is usually divided into octave or third-octave bands, the levels of which are controlled by sliders. The position of

the sliders gives a 'graphic' indication of the shape of the response curve.
Guardband Spacing between tracks on a recorded tape.

Haas effect (28) The effect that determines the apparent direction of a source of sound. When the same sound is reproduced simultaneously from two or more sources it will appear to come from the nearest one (ie with the minimum time lag) unless another source is much louder. The relationship between sound volume, time delay and directionality is known as the Haas effect.
Harmonic (14) A sinusoidal oscillation having a frequency which is an integral multiple of the fundamental frequency. It is the harmonics (or in musical terms *upper partials*) that shape the waveforms and make it possible to distinguish between various instruments, even when they are playing the same note.
Harmonic distortion (14) The production of spurious harmonics due to a distortion of the original waveform.
Hertz (Hz) (12) Unit of frequency: one hertz equals one cycle per second.
Hypercardioid (31) A directional response between cardioid and figure-of-eight, ie a rather narrower front lobe than cardioid but with a small lobe at the back. It is the optimum characteristic for discrimination between axial sound and all-round ambience.

Indirect sound (18) Sound which reaches the listener or microphone by acoustic reflection.
Intensity of sound (14) A measure of the power of a sound, measured in decibels relative to the threshold of hearing at 1000 Hz. Sound intensity is not the same as loudness (which is measured in *phons*) except at 1000 Hz because of the unequal frequency response of the ear. (See Phon.)
Intermodulation distortion (14) A form of distortion caused by one component frequency modulating another thereby creating spurious sum and difference frequencies which produce a 'rough' tone.
Leader (120) A section of uncoated tape, usually coloured white, which is joined to the beginning of a recording to allow for threading to the take-up spool.
Level The intensity of steady tone used for test purposes relative to a standard reference level. Zero level is normally taken as a power of 1 mW in a resistance of 600 Ω.
Limiter (60) A device for preventing the volume of a signal from rising above a pre-set value, thereby preventing overload. The action is similar to a compressor except that the gain reduction is more severe. A compression ratio of 10:1 or more can be considered as limiting as a very large increase input is required to make a significant difference to the output.

Loudness (14) The subjective impression of the strength of a sound. Loudness is affected by a number of factors such as the actual volume of the sound, the listener's aural sensitivity, the masking effect of one source of sound on another of similar frequency, and an 'irritation factor'. Unwanted sound (noise) tends to sound louder than wanted sound of similar volume.

Masking (66) The manner in which the ability to hear sounds of a particular frequency becomes reduced in the presence of louder sounds of a similar frequency.
Modulation The control of one waveform by another.
Monophonic (mono) The reproduction of sound via a single medium of transmission.
Mumetal (100) An iron alloy used for tape heads and for magnetic screening.

NARTB (or NAB) (110) Standard of National Association of Radio and Television Broadcasters. There are agreed NAB standard frequency characteristics for the recording and reproduction of discs and tapes.
Neopilaton A system of sycnhronising tape recordings used mainly for location film recording. Two narrow tracks are recorded by the camera, or by a stable oscillator, in opposite phase down the centre of the tape. The tracks can be used for synchronising but cancel out in the full-width replay used for programme.

Pan (48) To change direction in the lateral sense.
Parametric equalizer (56) A frequency-response shaping device in which the frequency and the bandwidth of the filter can be varied.
Peak programme meter (PPM) (64) A meter for measuring programme volume which has a rapid rise time and slow recovery so that it registers peak power.
Permeability (80) A measure of the ease of magnetisation of a material.
Perspective (16) The apparent effect of distance. The distance of a sound source is suggested by the relative volume level, the ratio of direct to reverberant sound, and the characteristic quality.
Phase The position in the cycle that a waveform has reached at any given instant. Waves are said to be in phase when their cyclic positions coincide.
Phon (12) A measure of the loudness of a sound that takes into account the unequal frequency response of our ears. Phons and decibels are the same at 1000 Hz. At other frequencies their relationship is illustrated by the curves of equal loudness.
Pitch (14) Subjective effect of sound related mainly to frequency but affected also by intensity and harmonic structure. As volume is increased, high sounds can seem higher and low sounds lower.

Pre-emphasis (70, 211) Boosting the high frequency component of a signal before recording or transmission so as to reduce the high frequency system noise when it is subsequently *de-emphasised* on reproduction.
Presence filter A filter which imparts a rise, usually in the form of a steep peak, to the frequency response of a circuit in the region of 3–7 kHz which is the frequency range in which the sibilants in speech tend to lie. The effect of closeness or 'presence' is given and clarity improved.
Print through (84) The transfer of a recording on to an adjacent layer when the tape is wound on a spool. Most noticeable when a high-level passage in one layer coincides with a quiet passage on an adjacent one. Print through is minimised by storing tape under conditions of moderate temperature and low humidity and by periodic re-spooling.

Quadraphony (50) A four-channel transmission/reproducing system intended to recreate a sound field that surrounds the listener.

Radio microphone (38) A microphone (usually a lavalier or hand-held type) equipped with a miniature transmitter to enable the artist to carry it about without the encumbrance of a cable.
Recovery time (60) The time taken for a compressor or limiter to restore the gain to normal on reduction of the signal.
Reverberation (20, 26) The sustaining effect of multiple sound reflections within an enclosure.
Reverberation time (20) The time taken for a sound to die away through 60 dB (one millionth of its original intensity) in a reverberant enclosure.
Remanence (78) The ability of a material to retain residual magnetisation when the magnetizing force has been removed or reversed.
Rumble (142) Low frequency vibration.

Signal-to-noise ratio The ratio of the magnitude of the programme signal to the system noise.
Splicing tape (120) Contact-adhesive tape used for butt-joining tapes when editing.
Squirrel-cage motor An induction motor in which the rotor is made of a series of copper rods terminating in circular end-rings giving the appearance of a cage.
Stereophony (48) A two-channel transmission/reproduction system providing the illusion of spatial distribution of the sources in the horizontal plane.
Stylus (146) The cutting head, usually tipped with sapphire or diamond, in a disc recorder or the playing needle in a pick-up cartridge.

Tape transfer characteristic (90) The relationship between the applied magnetising force (H) and the induced magnetic flux (B) in a recording tape.
Time constant (110) A method of stipulating the shape of a response curve by reference to the charge/discharge time of a capacitance and resistance in series.

Uni-directional microphone (34, 40) A microphone which is sensitive to sound from one direction only.

VU meter A meter for indicating programme volume which indicates signal power (in decibels) on steady tone and volume units (percentage utilisation of the channel) on programme.

Wavelength (12) The distance between corresponding parts of a sinusoidal waveform.
White noise A full-spectrum signal having the same energy level at all frequencies.
Windshield (38) A device fitted around a microphone to protect it from wind or air turbulence due to rapid movement through the air, eg on the end of a boom. Windshields have to be large to be effective against wind. Smaller close-talking shields are available for some microphones to guard against breath puffs.
Wow (112) A slow variation of pitch, normally most noticeable on low notes in recordings and due to speed instability in the recording or reproducing mechanism.

Zero level The level used for lining up equipment and the broadcasting chain using standard tone. It corresponds to a power of 1 mW in 600 Ω and is indicated by the figure 4 on a peak programme meter.